Multiserver 3000

User's Manual

Part Number 800-1856-50 Rev. A

May 1996

Safety Warnings and Cautions

Various safety agencies request statements of warning or caution to help you in the safe operation of the Black Box unit. These statements also apply to any and all modules installed within the unit.

To ensure adequate cooling of the equipment a 2.0 inch unobstructed space must be maintained around all sides of the unit.

The ac power socket shall be installed near the equipment and shall be easily accessible.

Installation and access to the interior of this unit shall be made only by a qualified technician.

Connection to the network is to be disconnected before the (mains) plug is removed.

Warning

Remove power plug from the power socket before performing any service work on the unit.

Lithium Battery Caution

Danger of explosion if battery is incorrectly replaced. Replace only with the same type or equivalent battery, as recommended by the manufacturer. Discard used batteries according to manufacturer's instructions.

The power supply is auto-ranging in this model.

The power supply cordset to be supplied in Europe must have 0.75^2mm, 3 conductor "HAR" cord type H05VV-F, terminated in a grounding type Shucko plug on one end and a molded-on IEC 320 connector on the other end.

Technical Data

Input Volts	:	100-240 Vac −5%, +10%
Input Current		
Marathon 5K Turbo, 10K, 20K		
NetRunner 500ET, 1000E, 2000E	:	3A-1.5A
Marathon 2K, 2K *Plus*, 3K		
NetRunner 75E	:	2A-1A
STADIA Integration Hub	:	6A-3A
Frequency	:	47-63 Hz

Um die Kühlung des Gerätes nicht zu beschränken, ist es notwendig um das Gerät herum an allen Seiten ca 5 cm Raum zu lassen.

Stellen Sie das Gerät in der Nähe einer geerdeten Schutzkontaktsteckdose so auf, dass diese leicht erreichbar und zugänglich ist.

Die Montage und der Zugang ins Innere des Gerätes sind nur einem qualifizierten Techniker gestattet.

Ehe der Netzstecker aus der Steckdose gezogen wird, müssen sämtliche äusserliche Verbindungen vom Gerät getrennt werden.

Warnung

Vor öffnen des Gerätes, muss der Netzstecker aus der Steckdose gezogen werden.

Lithium Batterie Warnung

Explosionsgefahr besteht wenn die Batterie nicht richtig ersetzt ist. Die Batterie darf nur mit einer gleichen oder gleichwertigen Batterie ersetzt werden.

Netzteil ist mit automatischer Umschaltung entsprechend der Versorgungsspannung versorgt.

Die Netzleitung sollte ein harmonisierter Typ (HAR) sein, mit der Bezeichnung H05VV-F oder H05VVH2-F, 3G 0.75^2mm, mit einem Schutzkontakt – und einem Kaltgerätestecker (IEC 320).

Technische Daten

Nennspannung	:	100-240 V∼ −5%, +10%
Nennstrom		
Marathon 5K Turbo, 10K, 20K		
NetRunner 500ET, 1000E, 2000E	:	3A-1.5A
Marathon 2K, 2K *Plus*, 3K		
NetRunner 75E	:	2A-1A
STADIA Integration Hub	:	6A-3A
Frequenz	:	47-63 Hz

Pour assurer un refroidissement adéquat, maintenir un espace libre de 5 cm (2 pouces) tout autour de l'appareil.

Installer la prise AC à proximité de l'appareil, dans un rayon d'accès facile.

L'installation et l'ouverture de cet appareil est permise par un technicien autorisé seulement.

Avant de débrancher la prise de courant, assurer que toutes les connexions externes ont été déconnecté de l'appareil.

Avertissement

Débrancher la prise de courant avant d'entreprendre aucun travail de réparation de l'appareil.

Batterie Au Lithium Avertissement

Un danger d'explosion existe si la batterie est remplacée incorrectement. Remplacer avec une batterie identique ou similaire, recommandée par le fabriquant. Disposer des batteries utilisées selon la méthode prescrite par le fabriquant.

Ce modèle s'adapte automatiquement au courant électrique ou voltage de la prise murale.

En Europe, brancher l'appareil à la prise murale au moyen d'un fil "HAR" comprenant 3 cables H05VV-F ou H05VVH2-F de 0.75^2mm chacun, avec à une extremité une prise de terre genre SHUCKO et à l'autre une prise IEC 320.

Donnees Techniques

Voltage d'Accès	:	100-240 V∼ −5%, +10%
Courant d'Accès		
Marathon 5K Turbo, 10K, 20K		
NetRunner 500ET, 1000E, 2000E	:	3A-1.5A
Marathon 2K, 2K *Plus*, 3K		
NetRunner 75E	:	2A-1A
STADIA Integration Hub	:	6A-3A
Fréquence	:	47-63 Hz

Notification of FCC Requirements

NOTE: This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at his own expense.

Changes or modifications to this product, that could increase the amount of Radio Frequency Emissions from this product, without the expressed written approval of Black Box Corp., could cause the product and the user to violate the FCC's Rules and Regulations, thus requiring the product to be turned off or disconnected.

If this unit is used on a DTE which requires use of shielded cables for compliance with FCC Part 15, then use of a filtered pin connector may be required to maintain FCC compliance. See the Installation section for specific applications.

Notification of Canadian Requirements

This digital apparatus does not exceed the Class A limits for radio noise emissions from digital apparatus as set out in the Radio Interference Regulations of the Canadian Department of Communications.

Le présent appareil numérique n'émet pas de bruits radioélectriques dépassant les limites applicables aux appareils numériques de classe A prescrites dans le règlement sur le brouillage radioélectrique édicté par le Ministère des Communications du Canada.

United Kingdom Requirement: Interconnection of Ports Warning

Interconnection directly, or by way of other apparatus, of ports marked "SAFETY WARNING. See instructions for use", with ports marked or not so marked may produce hazardous conditions on the network. The advice of a competent engineer must be obtained before such a connection is made. None of the ports provide isolation sufficient to satisfy the relevant parts of BS 6301. Apparatus connected to the ports, must either have been approved to the relevant parts of BS 6301 or to have been previously evaluated against BS 6301 British Telecom Technical Guides 2 or 26, and given permission to attach. Other usage will invalidate any approval given to this apparatus.

Any or all of the ports on the following modules may be configured as non-network ports:

- Communications Control Modules (CCM): 6 D-type ports
- Up to 4 Channel Expansion Modules (CEM): up to 6 D-type or 12 RJ45 ports
- LAN modules (including RTS, RLB, or IRM): AUI, BNC, and 8-pin modular jack ports
- NMS module: 1 log port, 1 command port
- Up to 4 2-port voice cards
- Alarm port

WARRANTY

Black Box warrants that to the extent that the equipment delivered is hardware, such equipment shall be free from defective material and workmanship for a period of 3 years from the date of shipment of equipment from Black Box when given normal, proper and intended usage. Black Box further agrees to provide, without cost, emergency replacement equipment, shipped freight prepaid, for a period of ninety (90) days from date of shipment of the equipment and factory repair for the remainder of the warranty period provided that:

(a) Black Box is promptly notified upon discovery that the equipment is defective;

(b) The equipment is returned freight prepaid to BlackBox;

(c) Black Box's examination of the equipment shall disclose that any defect was not caused by failure of electrical power or air conditioning, damage from lightning or weather-related causes, accident, misuse, neglect, alteration, improper installation, unauthorized repair or improper testing.

To the extent the equipment is or contains software or firmware (collectively "Software"), Black Box warrants that for a period of one (1) year from the date of shipment, the Software shall be free from defects in material and workmanship under normal use and that the programs will perform according to the specifications contained in BlackBox's user manual. BlackBox does not warrant that the functions contained in the Software will meet a specific requirement or that the operation will be uninterrupted or error free.

INSTALLING NON BLACKBOX SOFTWARE IN BLACKBOX EQUIPMENT SHALL VOID THIS WARRANTY.

Black Box may, in its sole discretion, except for the first ninety (90) days of warranty, elect to repair or replace the equipment, in which event Black Box shall have a reasonable time to make repairs or to replace the equipment. Black Box will return the equipment freight prepaid.

THE PROVISIONS OF THIS WARRANTY ARE IN LIEU OF ANY OTHER WARRANTY, WHETHER EXPRESS OR IMPLIED, WRITTEN OR ORAL (INCLUDING ANY WARRANTY OF MERCHANTABILITY OR FITNESS FOR A PARTICULAR PURPOSE), AND Black Box LIABILITY ARISING OUT OF THE MANUFACTURE, SALE, OR SUPPLYING OF THE EQUIPMENT OR ITS USE, WHETHER BASED UPON WARRANTY, CONTRACT, NEGLIGENCE, PRODUCTS LIABILITY OR OTHERWISE, SHALL NOT EXCEED THE ORIGINAL AMOUNT PAID BY THE BUYER FOR THE EQUIPMENT. IN NO EVENT SHALL BLACK BOX BE LIABLE TO THE BUYER OR ANY OTHER PERSON OR ENTITY FOR UNINTENDED OR CONSEQUENTIAL DAMAGES (INCLUDING, BUT NOT LIMITED TO, LOSS OF PROFITS OR USE DAMAGES) ARISING OUT OF THE MANUFACTURE, SALE OR SUPPLYING OF THE EQUIPMENT.

SERVICE INFORMATION

If you experience difficulty with this product, contact your BLACK BOX Certified Distributor for prompt assistance.

BLACK BOX offers complete factory repair for both in-warranty and out-of-warranty equipment.

Before returning any equipment, you *must* obtain a Return Authorization number. Contact your distributor for assistance.

Shipping charges must be prepaid.

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1 General Description

This manual describes the dual-channel voice/fax portion of the 3K Base Module.

Physical Description

Figure 1-1 is a front top view of the 3K Base Module. It is made up of two distinct parts: the dual channel Voice/Fax Module, and the Communications Control Module (CCM). As viewed from the front, all circuits associated with voice/fax occupy the left half of the module, while the Communications Control Module is on the right side. Since this manual deals mainly with voice/fax functionality, the Communications Control Module will be discussed only where it directly affects voice/fax operations.

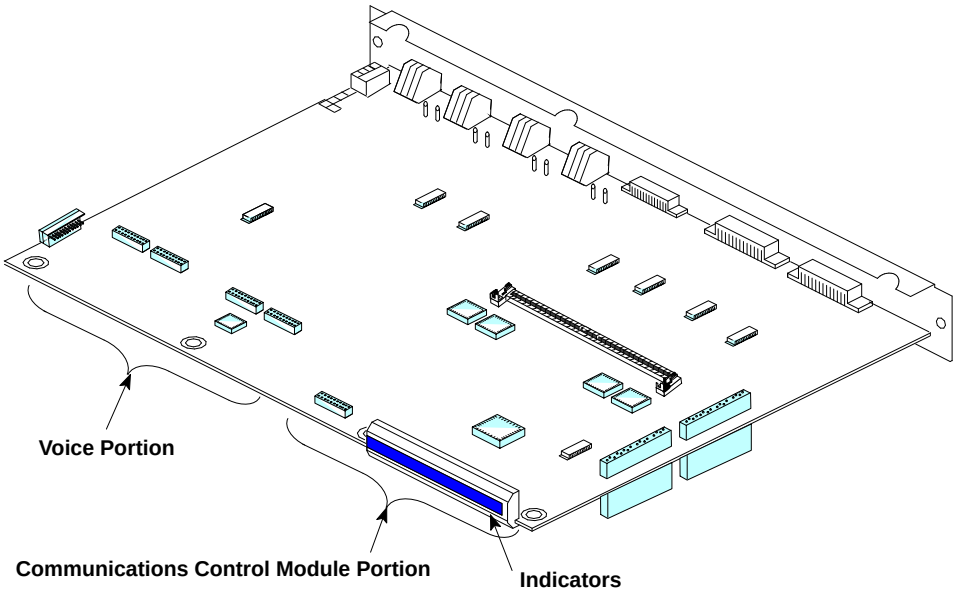

Figure 1-1. 3K Base Module, Front Top View

On the right front of the module are the 14 indicators. Of these, the left eight are associated with the two voice/fax channels, while the last six are related to the Communications Control Module. The voice/fax indicators will be discussed in more detail in Sections 5 and 6.

Figure 1-2 is the rear view of the 3K Base Module. Each voice/fax channel has a pair of connectors associated with it: RJ-1CX and RJ11. These are used to connect the voice/fax ports to the telephone interface.

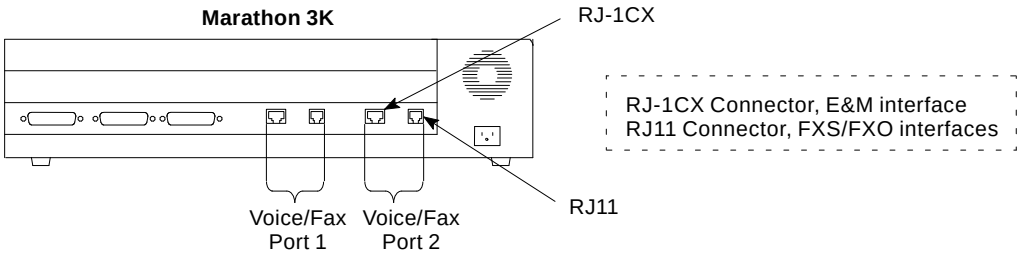

Figure 1-2. 3K Base Module, Rear View

Voice/Fax Operation

Each voice/fax channel converts analog voice obtained from a telephone interface into digital form, and internally connects the converted voice to the Communications Control Module portion of the 3K Base Module (see Figure 1-3). Next, the converted digitized signal is multiplexed with other data to form part of the Communications Control Module composite data stream. This data stream is transmitted over a data link to a remote Communications Control Module.

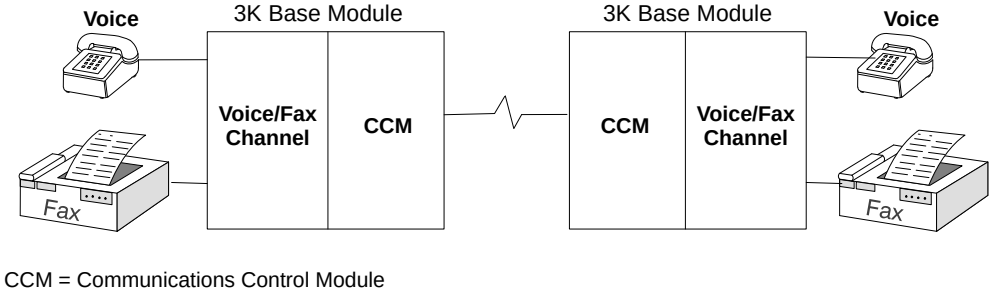

Figure 1-3. Voice/Fax Operation

At the remote Communications Control Module the digitized voice signal is demultiplexed, and routed to a compatible voice/fax channel specified by the destination address. Ultimately, the digitized voice signal is reconverted to the original analog form.

In addition to accepting analog voice signals for digital conversion, the voice/fax channel can also process analog signals obtained from a fax machine or modem. This feature is implemented by resident firmware.

The firmware constantly monitors the incoming analog stream for the presence of fax or modem signals. When fax or modem signals are detected, and provided there is currently no voice traffic on the channel, the firmware switches into the fax/modem mode. While in this mode, the signals are demodulated into the appropriate fax or modem symbols and passed as data to the remote end. At the remote end the fax or modem symbols are remodulated to the correct standards.

Telephone Interface Types

Depending on the telephone interface, the voice/fax facility can be strapped to conform with one of three common types of signaling convention:

- E&M (Ear and Mouth) – A tie line trunk circuit used to connect between Private Branch Exchanges (PBXs) or other voice switching system.
- FXO (Foreign Exchange Office) – A trunk loop start operation that emulates a single-line telephone to Central Office lines or CENTREX® or PBX stations. It recognizes ringing signals and draws current to indicate an active state.
- FXS (Foreign Exchange Station) – A station loop start operation that provides connection to a standard, single-line telephone instrument, the line circuit of a Key Telephone System (KTS), or a loop start trunk circuit of a PBX that normally connects to incoming Central Office (CO) circuits. This interface type provides power and ringing signals to its interfacing equipment. It is not intended for connection to the Public Switched Telephone Network (PSTN).

Strapping

All information relating to the strapping of the voice/fax facility interface types is provided in the 800-1846 3-Slot Chassis Installation and Cabling Manual.

Connecting the Voice/Fax Channel to the Telephone Equipment

All information relating to the telephone connection, including interface cabling and pinouts, is provided in the 800-1846 3-Slot Chassis Installation and Cabling Manual.

Compatibility

The 3K voice is compatible with other BLACK BOX analog voice modules. The voice module with the least features will determine the feature set for the channel.

Related Documentation

- Marathon 3-Slot Chassis Installation and Cabling Manual, BLACK BOX
- Marathon Startup Guide,
- Universal Voice/Fax Module and Telephone Interface Modules User's Manual

Configuration 2

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This section describes how to configure the voice/fax channel. Your Startup Guide explains how to access the Command Facility Main Menu.

Menu Structure

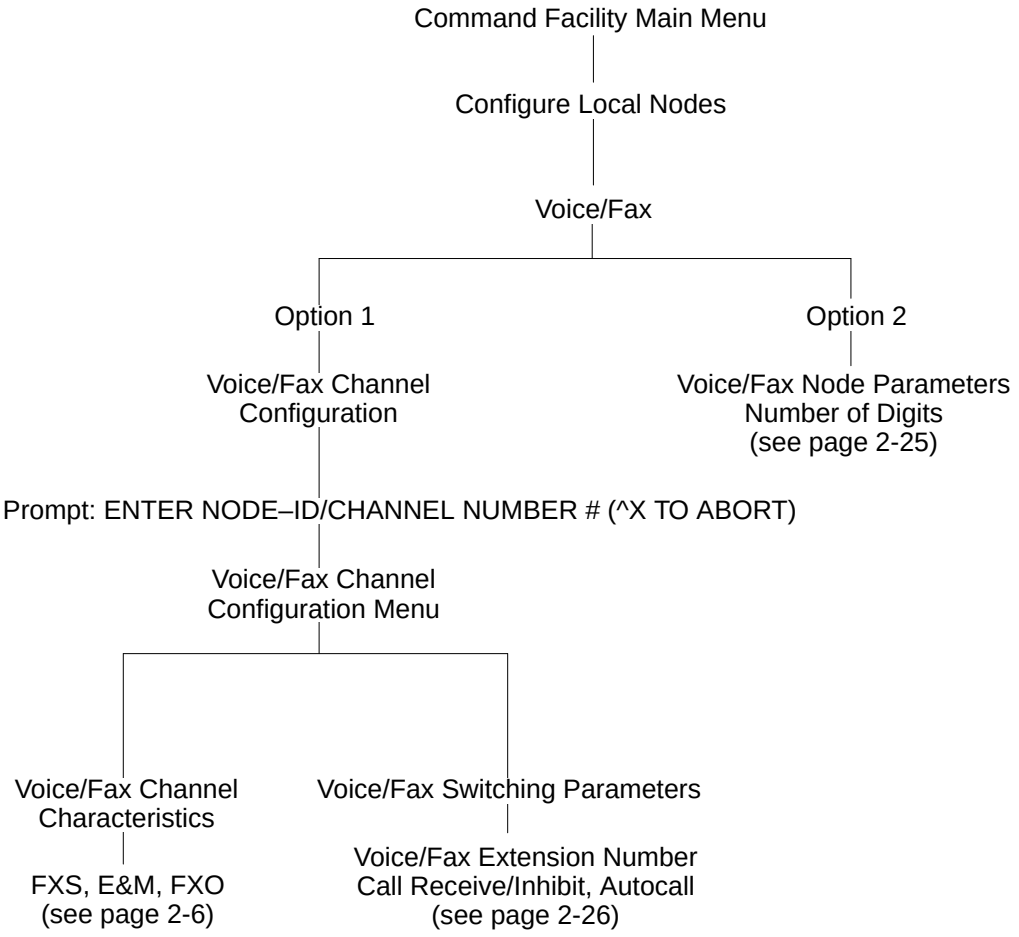

Command Facility Main Menu
Configure Local Nodes
Voice/Fax
Option 1
Option 2
Voice/Fax Channel Configuration
Voice/Fax Node Parameters Number of Digits (see page 2-25)
Prompt: ENTER NODE–ID/CHANNEL NUMBER # (^X TO ABORT)
Voice/Fax Channel Configuration Menu
Voice/Fax Channel Characteristics
Voice/Fax Switching Parameters
FXS, E&M, FXO (see page 2-6)
Voice/Fax Extension Number Call Receive/Inhibit, Autocall (see page 2-26)

Figure 2-1. Voice/Fax Configuration Outline

Summary of Options

Tables 2-1 through 2-3 summarize the configuration options associated with the voice/fax channel. Refer to pages 2-7 through 2-8 for option numbers of each interface type.

Table 2-1. Voice/Fax Channel Characteristics Options

Option	Telephone Interface	Default	Description (Page Reference)
Mode	All	Voice/Fax	Selects two types of channel signal processing: voice/fax, and voice only (page 2-8).
Digitizing Rate	All	G.729	Selects one of ten voice digitizing rates (page 2-9).
Input Level Gain	All	0 dB	Selects one of 26 input signal gain values in 1-dB increments; -6 is maximum attenuation; 0 = no gain; 19 = maximum gain (page 2-10).
Output Level Attenuation	All	0 dB	Selects one of 26 output level attenuation values in 1-dB increments; 0 = no attenuation; 25 = maximum attenuation (page 2-10).
Busyout Mode	All	System Controlled	Selects one of three methods of busyout control (page 2-11).
Bandwidth	All	Voice Activated	Selects how the channel uses link bandwidth: all the time, only when voice or fax signals are present, or dynamically (page 2-12).
Background	All	Regenerated	Selects to either regenerate or suppress background noise during idle periods. (page 2-13).
Priority	All	High	Selects either high or low composite bandwidth priority for the channel (page 2-13).
Signalling Format	E&M only	DC	Selects one of three types of signaling format: DC, where signaling is accomplished by the state of the E and M leads; 2280, Tone, where signaling is accomplished by tone over the analog transmission path; and pulsed DC, where signaling follows Ron/Tron convention (page 2-14).
Signalling Format	FXS only	Interrupted Ring 2/4	Selects one of two ringing patterns (page 2-14).

Table 2-1. Voice/Fax Channel Characteristics Options (continued)

Option	Telephone Interface	Default	Description (Page Reference)
Number of Rings	FXO only	1	Selects number of ring backs before receipt of dial tone (page 2-15).
Analog Operation	E&M only	4-wire	Sets channel for either 2-wire or 4-wire operation (page 2-16).
Ringing Frequency	FXS only	25 Hz	Selects 25 Hz or 50 Hz ringing frequency (page 2-16).
Fax Digitizing Rate	All	Voice Rate	Selects 1 of 5 digitizing rates for fax operation (page 2-17).
Disconnect Supervision	FXO only	Tone	Selects either tone or power interrupt disconnect supervision (page 2-18).
Line Impedance	FXS, FXO, and E&M 2-wire only	600 ohms	Selects one of two line impedance matching types (page 2-19).
Maximum Output Level	E&M 4-wire only	0 dBm	Selects one of two maximum output levels (page 2-20).
Regeneration Delay	All	1	Delays from 1 to 15 seconds the forwarding of dial digits to the destination PBX (page 2-20).
Dial Digit Time Limit	All	10	Selects from 1 to 15 seconds of interdigit time limit. When time limit is reached, the voice/fax channel stops detecting dialing digits (page 2-21).
Maximum Number of Forwarded Digits	All	0	Ranges from 1 to 16. When the configured number is reached, the digits are forwarded without waiting for the time limit (page 2-22). 0 disables the single dial tone feature.
Regeneration Format	All	Dial Pulse	Specifies 1 of 2 types of regeneration format, pulse or DTMF (page 2-23).
Call Progress Tone	All	North American	Specifies 1 of of 8 types of call progress tones, as required in selected countries or regions (page 2-23).

Table 2-2. Voice/Fax Node Parameters Option

Option	No.	Telephone Interface	Default	Description
Number of Digits	1	FXS E&M FXO	2	Specifies number of digits (1-4) to be assigned to each voice/fax extension (page 2-25).

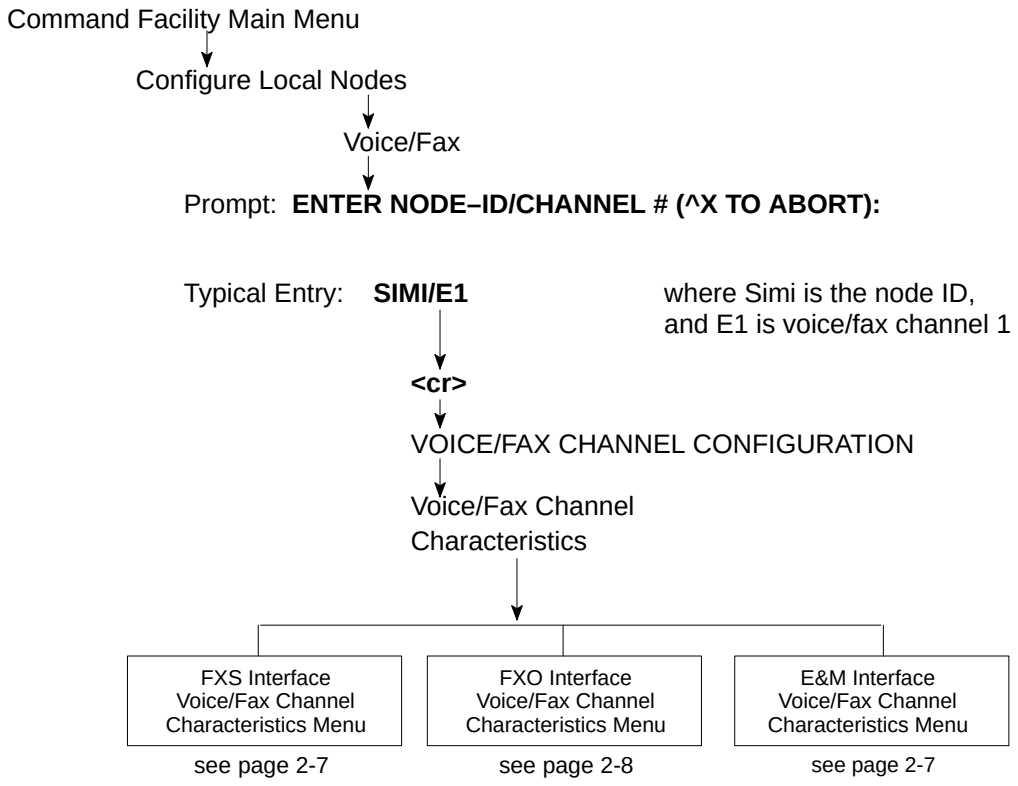
Table 2-3. Voice/Fax Switching Parameters Options

Option	No.	Telephone Interface	Default	Description
Voice Extension Number	1	FXS E&M FXO	–	Assigns the telephone extension number to the voice/fax channel (page 2-27).
Fax Extension Number	2	Not used in the Marathon 3K	–	
Call Inhibit	3	FXS E&M FXO	NO	When set to YES, prevents the voice/fax channel from calling another voice/fax channel (page 2-27). It can only answer calls.
Receive Inhibit	4	FXS E&M FXO	NO	When set to YES, prevents the voice/fax channel from receiving calls (page 2-27). It can only originate calls.
Autocall Extension Number	5	FXS E&M FXO	–	When off-hook, automatically calls the configured extension number (page 2-27).

Configuring the Channel Characteristics

Accessing the Menus

To access the channel characteristics menus, access the Command Facility Main Menu, then follow this path.


Command Facility Main Menu
Configure Local Nodes
Voice/Fax
Prompt: ENTER NODE–ID/CHANNEL # (^X TO ABORT):
Typical Entry: SIMI/E1
where Simi is the node ID, and E1 is voice/fax channel 1
<cr>
VOICE/FAX CHANNEL CONFIGURATION
Voice/Fax Channel Characteristics
FXS Interface Voice/Fax Channel Characteristics Menu
see page 2-7
FXO Interface Voice/Fax Channel Characteristics Menu
see page 2-8
E&M Interface Voice/Fax Channel Characteristics Menu
see page 2-7

The menu you will see depends on the type of interface you have.

FXS Interface

```
VOICE/FAX CHANNEL CHARACTERISTICS [node ID/channel #]
  1. MODE                                        [VOICE/FAX]
  2. DIGITIZING RATE                               [G.729]
  3. INPUT LEVEL GAIN                              [0 dB]
  4. OUTPUT LEVEL ATTENUATION                      [0 dB]
  5. BUSYOUT MODE                  [SYSTEM CONTROLLED]
  6. BANDWIDTH                       [VOICE ACTIVATED]
  7. BACKGROUND                        [REGENERATED]
  8. PRIORITY                                      [HIGH]
  9. SIGNALLING FORMAT               [INTERRUPTED RING]
 10. RINGING FREQUENCY                             [25 HZ]
 11. FAX DIGITIZING RATE                    [VOICE RATE]
 12. LINE IMPEDANCE                          [600 OHMS]
 13. REGENERATION DELAY                             [1]
 14. DIAL DIGIT TIME LIMIT                          [10]
 15. MAXIMUM NUMBER OF FORWARDED DIGITS             [0]
 16. REGENERATION FORMAT                    [DIAL PULSE]
 17. CALL PROGRESS TONE                       [N. AMER]

CR - ACCEPT ENTRY
M  - MAIN MENU
```

Note: The information in brackets shown in the menus are the default values. As you configure your unit, the information displayed will be updated to the current values.

E&M Interface

```
VOICE/FAX CHANNEL CHARACTERISTICS [node ID/channel #]
  1. MODE                                        [VOICE/FAX]
  2. VOICE DIGITIZING RATE                         [G.729]
  3. INPUT LEVEL GAIN                              [0 dB]
  4. OUTPUT LEVEL ATTENUATION                      [0 dB]
  5. BUSYOUT MODE                  [SYSTEM CONTROLLED]
  6. BANDWIDTH                       [VOICE ACTIVATED]
  7. BACKGROUND                        [REGENERATED]
  8. PRIORITY                                      [HIGH]
  9. SIGNALLING FORMAT                             [DC]
 10. ANALOG OPERATION                           [4-WIRE]
 11. FAX DIGITIZING RATE                    [VOICE RATE]
 12. LINE IMPEDANCE (2-WIRE ONLY)            [600 OHMS]
 13. MAXIMUM OUTPUT LEVEL (4-WIRE ONLY)         [0 DBM]
 14. REGENERATION DELAY                             [1]
 15. DIAL DIGIT TIME LIMIT                          [10]
 16. MAXIMUM NUMBER OF FORWARDED DIGITS             [0]
 17. REGENERATION FORMAT                    [DIAL PULSE]
 18. CALL PROGRESS TONE                       [N. AMER]

CR - ACCEPT ENTRY
M  - MAIN MENU
```

Note: The information in brackets is the default value. As you configure your unit, the information displayed will be updated to the current values.

FXO Interface

```
VOICE/FAX CHANNEL CHARACTERISTICS [node ID/channel #]
  1. MODE                                        [VOICE/FAX]
  2. DIGITIZING RATE                               [G.729]
  3. INPUT LEVEL GAIN                              [0 dB]
  4. OUTPUT LEVEL ATTENUATION                      [0 dB]
  5. BUSYOUT MODE                  [SYSTEM CONTROLLED]
  6. BANDWIDTH                       [VOICE ACTIVATED]
  7. BACKGROUND                        [REGENERATED]
  8. PRIORITY                                      [HIGH]
  9. NUMBER OF RINGS                               [1]
 10. FAX DIGITIZING RATE                 [VOICE RATE]
 11. DISCONNECT SUPERVISION                        [TONE]
 12. LINE IMPEDANCE                          [600 OHMS]
 13. REGENERATION DELAY                           [1]
 14. DIAL DIGIT TIME LIMIT                        [10]
 15. MAXIMUM NUMBER OF FORWARDED DIGITS           [0]
 16. REGENERATION FORMAT                  [DIAL PULSE]
 17. CALL PROGRESS TONE                      [N. AMER]

CR - ACCEPT ENTRY
M  - MAIN MENU
```

Note: The information in brackets is the default value. As you configure your unit, the information displayed will be updated to the current values.

Description of Options

Mode

The MODE option of the Channel Characteristics Menu allows you to select two types of voice/fax operation.

```
MODE [VOICE/FAX]
        1. VOICE/FAX
        2. VOICE ONLY

CR - ACCEPT ENTRY
M  - MAIN MENU
N  - NEXT ENTRY

ENTRY:
```

If you select the VOICE/FAX option, the channel will process voice or fax signals, as follows: Normally the channel operates in the voice mode. If a fax signal is detected, and provided there is no voice traffic over that channel, the channel will automatically switch to the fax mode. It will stay in that mode until fax signals are no longer detected, and then switch back to the voice mode. Use this option for most voice/fax applications.

If you select the VOICE ONLY option, the voice/fax channel will be in the voice mode all the time. Use this as the preferred option if the channel is to be dedicated to voice traffic.

Note: When force connected, both local and remote channels must be configured for the same mode. When attempting a switched call, if either channel is set to Voice Only, that call will operate in the Voice Only mode.

Digitizing Rate

The DIGITIZING RATE option of the Channel Characteristics Menu allows you to select one of 10 voice digitizing rates in bps as shown in the following display. Higher voice digitizing rates will usually produce crisper voice communications. However, higher digitizing rates may result in overbooking of composite bandwidth.

```
DIGITIZING RATE          [G.729]
        1.   4000
        2.   4800
        3.   6400
        4.   7200
        5.   8000
        6.   9600
        7.  12000
        8.  14400
        9.  16000
       10.  G.729

CR - ACCEPT ENTRY
M  - MAIN MENU
N  - NEXT MENU

ENTRY:
```

The G.729 option, part of the ClearVoice Technology, is incorporated in Release 5.0 or later Marathon 3K units. It uses a voice compression algorithm based on International Telecommunications Union's (ITU) G.729 recommendations. With G.729, voice is digitized at 8 Kbps, but produces voice quality equivalent to 32-Kbps voice using Adaptive Differential Pulse Code Modulation (ADPCM).

If you call a network extension that does not have the G.729 option, the call will reduce to the lowest rate and utilize the older voice algorithm. See Table 2-4 for compatibility details.

When in the force connect mode, you may use the G.729 option only when both sides are set to G.729.

Bandwidth Selection

With the voice rate of a voice channel configured for G.729 rate, use VOICE ACTIVATED only. If you select DYNAMIC or CONTINUOUS, the message "NOT SUPPORTED" will appear.

Table 2-4. Compatibility Matrix

Node A		Node B		
Voice/Fax Channel	**Rel. Level**	**Voice/Fax Channel**	**Rel. Level**	**Compatibility**
Configured for G.729 rate	5.0	Configured for G.729 rate	5.0	Fully compatible, operating with G.729 algorithm.
Configured for G.729 rate	5.0	G.729 or non-G.729, configured for 4 Kbps to 7.2 Kbps	5.0	Operates with older voice algorithm at a lower rate.[1]
Configured for G.729 rate	5.0	G.729 or non-G.729, configured for 8 Kbps to 16 Kbps	5.0	Operates with older voice algorithm at 8 Kbps.[1]
Configured for G.729 rate	5.0	G.729 or non-G729 rate	4.3 or earlier	Operates with older voice algorithm at the lowest rate.[1]

[1] Will not operate in force connect mode

Input and Output Levels

The INPUT LEVEL option of the Channel Characteristics Menu provides one of 26 choices of input signal amplification settings in 1 dB increments as shown in the following display; –6 dB is maximum attenuation (negative gain), 0 is no amplification, and 19 dB is maximum amplification. The default is 0 dB.

Note: If your equipment is operated in the U.S.A. or Canada, input and output levels must be set to 0 dB. See Appendix B for details.

```
INPUT LEVEL GAIN  [0 dB]
  1.    -6 dB     14.    7 dB
  2.    -5 dB     15.    8 dB
  3.    -4 dB     16.    9 dB
  4.    -3 dB     17.   10 dB
  5.    -2 dB     18.   11 dB
  6.    -1 dB     19.   12 dB
  7.     0 dB     20.   13 dB
  8.     1 dB     21.   14 dB
  9.     2 dB     22.   15 dB
 10.     3 dB     23.   16 dB
 11.     4 dB     24.   17 dB
 12.     5 dB     25.   18 dB
 13.     6 dB     26.   19 dB

CR  -  ACCEPT ENTRY
M   -  MAIN MENU
N   -  NEXT MENU

ENTRY:
```

The OUTPUT LEVEL option of the Channel Characteristics Menu provides one of 26 choices of output signal attenuation settings in 1 dB increments as shown in the following display; 0 dB is no attenuation and 25 dB is maximum attenuation.

```
OUTPUT LEVEL ATTENUATION [0dB]
  1.    0 dB      14.   13 dB
  2.    1 dB      15.   14 dB
  3.    2 dB      16.   15 dB
  4.    3 dB      17.   16 dB
  5.    4 dB      18.   17 dB
  6.    5 dB      19.   18 dB
  7.    6 dB      20.   19 dB
  8.    7 dB      21.   20 dB
  9.    8 dB      22.   21 dB
 10.    9 dB      23.   22 dB
 11.   10 dB      24.   23 dB
 12.   11 dB      25.   24 dB
 13.   12 dB      26.   25 dB

CR  -  ACCEPT ENTRY
M   -  MAIN MENU
N   -  NEXT MENU

ENTRY:
```

Busyout Mode

The BUSYOUT MODE option of the Channel Characteristics Menu allows you to select one of three busyout modes.

```
BUSYOUT MODE [SYSTEM CONTROLLED]
        1.  SYSTEM CONTROLLED
        2.  FORCED ON
        3.  FORCED OFF

CR - ACCEPT ENTRY
M  - MAIN MENU
N  - NEXT ENTRY

ENTRY:
```

Selecting SYSTEM CONTROLLED will cause the system to place the channel in the busyout state in a force connect mode when there is no link bandwidth available for the voice/fax channel.

Selecting FORCED ON forces the voice/fax channel into the busyout state. This option is used to disable an intermittent or defective voice/fax channel until it can be repaired.

Selecting FORCED OFF prevents a force-connected channel from going busy, overriding any other existing conditions. Normally, this option is used when the channel is being tested.

Note that, when the channel is in the busyout mode, the LO indicator will flash every second.

Bandwidth

The BANDWIDTH option of the Channel Characteristics Menu allows you to select one of three methods of bandwidth utilization by the voice/fax channel.

```
BANDWIDTH [VOICE ACTIVATED]
        1. VOICE ACTIVATED
        2. CONTINUOUS
        3. DYNAMIC

CR - ACCEPT ENTRY
M  - MAIN MENU
N  - NEXT ENTRY

ENTRY:
```

Note:

- In a force connect mode, bandwidth must be set the same way at both ends. If changed at one end the connection will be disrupted until the other end is changed to match.
- When using the G.729 option, select VOICE ACTIVATED. Do not select the other options.

When VOICE ACTIVATED is selected, the channel uses link bandwidth only when speech or fax signals are present.

When CONTINUOUS is selected, link bandwidth is dedicated to the channel while that channel is active (off-hook), even during pauses in voice and fax signal transmissions; while active, the bandwidth of this channel is not available for other voice/fax, data, or LAN traffic. You may use this option if it is desired to transmit background noise, subject to availability of link bandwidth. If you use this option in pulsed dc E&M applications, then you will be using bandwidth even when the voice/fax channel is idle (similar to a TDM operation).

When DYNAMIC is selected, the voice/fax channel will normally operate at the configured digitizing rate, but will reduce to a lower rate when demand on composite link is heavy. This feature is effective over multiple hops, and is limited to inter-nodal delays of 60 milliseconds or less. As a general rule, use option 3 in networks of less than four nodes (three-node hops). If these limitations are exceeded, the dynamic voice option may cause voice dropouts during network congestion. This option is not recommended for operation over satellite links and for tandem circuits, and is not supported in BLACK BOX mux channels. Channels operating in the fax mode will not be down rated.

Background

The BACKGROUND option of the Channel Characteristics Menu provides you with two options for managing background noise.

```
BACKGROUND [REGENERATED]
        1.  REGENERATED
        2.  SILENCE

CR - ACCEPT ENTRY
M  - MAIN MENU
N  - NEXT ENTRY

ENTRY:
```

When REGENERATED is selected, background sound is reproduced locally and heard by the local telephone user.

When SILENCE is selected, gaps in speech are filled by silence. Try both options, then choose the preferred setting.

Priority

This option allows you to set a high or low bandwidth priority level for the selected voice/fax channel. This feature is primarily intended to establish priority between synchronous data channels and voice/fax channels during heavy demands on composite bandwidth. The default setting for every voice/fax channel is HIGH, and for every synchronous data channel is LOW.

```
PRIORITY [HIGH]
        1.  HIGH
        2.  LOW

CR - ACCEPT ENTRY
M  - MAIN MENU
N  - NEXT ENTRY

ENTRY:
```

Selecting one or more synchronous channels for high priority will allow those channels to contend equally with voice/fax channels for link bandwidth.

If the link bandwidth for the voice/fax channels is overbooked, then Dynamic Bandwidth is recommended. See page 2-12.

If you set the voice/fax channels for LOW priority and the synchronous channels for HIGH priority, then voice performance will clearly suffer, and Bandwidth should be configured to DYNAMIC to prevent voice breakup. However, voice breakup may still occur if the synchronous channels and minimum voice rates exceed the link bandwidth.

Signalling Format

FXS

```
FXS SIGNALLING FORMAT [INTERRUPTED RING]

        1. REPEATED RING
        2. INTERRUPTED RING

CR - ACCEPT ENTRY
M  - MAIN MENU
N  - NEXT ENTRY

ENTRY:
```

When REPEATED RING is selected, ringing depends on the ring cycle generated by the PBX. REPEATED RING must be used when the local channel is force connected to the remote channel, and the remote channel is strapped for FXO.

When INTERRUPTED RING is selected, it causes the local telephone to ring when the remote telephone is lifted off-hook, or vice versa. The ringing cadence is as follows:

ring for two seconds, off for four seconds.

INTERRUPTED RING is normally selected when an FXS-strapped voice/fax channel is connected to another FXS- or E&M-strapped voice/fax. INTERRUPTED RING *must* be selected for switched voice calls channels.

Note: INTERRUPTED RING selects the ringing cadence of the telephone set attached to the FXS-strapped voice/fax channel. It does not affect the cadence of the ring back heard by the user originating the call. That cadence is selected from the Call Progress Tones Menu as detailed on page 2-24.

E&M

```
E&M SIGNALLING [DC]

        1. DC
        2. 2280 TONE
        3. PULSED DC

CR - ACCEPT ENTRY
M  - MAIN MENU
N  - NEXT ENTRY

ENTRY:
```

When DC is selected, the channel senses the idle/active status by the DC condition of the M-lead, and sends the idle/active status to the remote channel as a DC condition of the E-lead.

When 2280 TONE, is selected, the channel senses the idle status by the presence of a 2280-Hz tone on the T-R pair, and holds a 2280-Hz tone as an idle condition on the T1-R1 pair. Option 2 supports CEPT-L1 circuits such as the AC15 circuits in the United Kingdom. Both ends of the circuit must support this signaling format and the channel must be force connected. This option can only be selected when the channel is configured for 4-wire analog operation (refer to page 2-16).

When PULSED DC is selected, going off-hook is indicated by a pulse of medium duration, and going on-hook is indicated by a pulse of a longer duration. Pulsed DC includes the Ron/Tron signaling convention. Both ends of the circuit must support this signaling format, and the channel must be force connected. In this mode, signals on the M lead are reproduced on the distant E lead and the voice/fax channel is not affected by the state of the signaling levels.

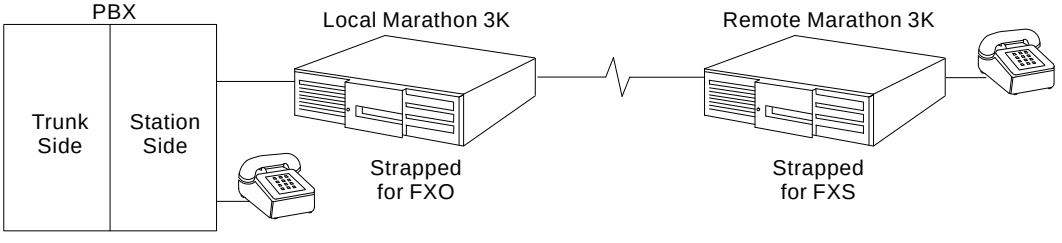
Number of Rings

```
NUMBER OF RINGS [1]
   1.    1
   2.    2
   3.    3
   4.    4
   5.    5
   6.    6
   7.    7
   8.    8
   9.    9

CR - ACCEPT ENTRY
M  - MAIN MENU
N  - NEXT MENU

ENTRY:
```

Application: In voice/fax switching, with the local channel fitted with an FXO interface module.



When an attempt is made to contact a network channel from a PBX station, the user first dials the local FXO station number and waits for a second dial tone. In response, the local FXO channel waits one or more ring cycles before providing that dial tone. Then the user dials the extension number of the network channel. It is the number of ring cycles before the dial tone that is selected with this option.

Notes:
- The NUMBER OF RINGS option applies *only* to calls *originating* at the FXO side of the WAN; it does *not* affect calls *received* by the FXO side.
- If the number of rings is configured for 8 or 9, the actual number of rings heard may be 9 or more.

Analog Operation

E&M

```
ANALOG OPERATION [4-WIRE]
        1.  2-WIRE
        2.  4-WIRE

CR - ACCEPT ENTRY
M  - MAIN MENU
N  - NEXT ENTRY

ENTRY:
```

Select the setting, as appropriate, to match the voice/fax channel with the associated PBX equipment.

Note: You must select the 4-WIRE option if you have selected the 2280 TONE signalling option on page 2-14.

Ringing Frequency

FXS

```
RINGING FREQUENCY [25 HZ]
        1.  25 HZ
        2.  50 HZ

CR - ACCEPT ENTRY
M  - MAIN MENU
N  - NEXT ENTRY

ENTRY:
```

This option controls the signal that rings the bell on the telephone to indicate that a call is coming to that location.

The 25 HZ option is normally used. However, some European telephone systems will only operate with a 50-Hz ringing frequency.

Fax Digitizing Rate

This option allows you to select one of five fax digitizing rates.

```
FAX DIGITIZING RATE (bps) [VOICE RATE]
        1.  VOICE RATE
        2.  2400
        3.  4800
        4.  7200
        5.  9600

CR - ACCEPT ENTRY
M  - MAIN MENU
N  - NEXT ENTRY

ENTRY:
```

The VOICE RATE option sets the fax digitizing to the same rate configured for voice (see page 2-9). The other four options (2 through 5) are used when voice and fax require different digitizing rates, typically when the voice rate is below 9600 bps. If you select VOICE RATE, and the voice digitizing rate is set for G.729 (see page 2-9), the fax signals will be digitized at 7.2 Kbps.

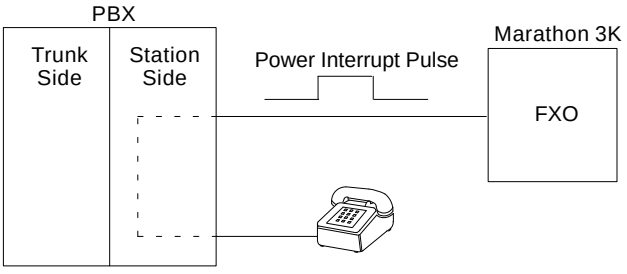
Manual fax switchover is not supported. You cannot send a fax message while a voice call is in progress; you must first set the fax machine or modem to auto answer and place a new call. The fax/modem detector relies on the answer tone of the called device. This tone is provided automatically when the called device is set for the auto answer mode.

Disconnect Supervision

Disconnect supervision is the means used to disconnect a switched telephone connection when the telephone equipment at one end goes from an off-hook (active) to an on-hook (idle) state.

Disconnect supervision is not effective in a forced telephone connection. However, Autocall can be used to accomplish a similar operation and provide disconnect supervision between an FXO and an FXS interface pair.

Voice/fax channels strapped for FXO can detect one of two types of disconnect supervision signals: power interrupt, or tone. In case of power interrupt, the voice/fax channel monitors the station line of the attached PBX or CO for the absence of loop current for 600 ms or longer.



If the attached PBX or CO cannot supply a power interrupt signal, the voice/fax channel looks for a call progress tone of 600 Hz or less.

```
DISCONNECT SUPERVISION [TONE]
        1. TONE
        2. POWER INTERRUPT

CR - ACCEPT ENTRY
M  - MAIN MENU
N  - NEXT ENTRY

ENTRY:
```

Select TONE if the attached PBX or Central Office (CO) cannot supply a power interrupt pulse. Select POWER INTERRUPT if the attached PBX or CO provides a 600-ms power interrupt pulse when the telephone set on the other side goes on-hook. This setting is recommended for CENTREX (Bell System) or CENTRANET (GTE) station lines from the CO.

Line Impedance (FXS, FXO, and E&M 2-Wire Only)

```
LINE IMPEDANCE [600 OHMS]
        1.  600 OHMS
        2.  COMPLEX

CR - ACCEPT ENTRY
M  - MAIN MENU
N  - NEXT ENTRY

ENTRY:
```

The LINE IMPEDANCE option provides impedance matching choices for the FXS, FXO, and E&M 2-wire interfaces as shown in Table 2-5. The line impedance of E&M interface configured for 4-wire operation is fixed at 600 ohms and is not configurable.

Table 2-5. Line Impedance Matching Options For Different Interface Modules

Option	Impedance	FXS	FXO	E&M 2-wire	E&M 4-wire
1. 600 Ω	600 Ω resistive	No	No	Yes	Fixed
	600 Ω + 2.2 μF	Yes	Yes	No	No
2. Complex	370 Ω + 0.31 μF // 620 Ω or 220 Ω + 0.12 μF // 820 Ω	Yes	Yes	Yes	No

The symbol // means in parallel with.

Maximum Output Level (E&M 4-Wire Only)

The +7 dBm option of this menu is used when the interfacing tie trunk equipment includes a pad. Use 0 dBm NOMINAL for all other applications.

```
MAXIMUM OUTPUT LEVEL [0 dBm NOMINAL]
        1.  0 dBm NOMINAL
        2.  +7 dBm

CR - ACCEPT ENTRY
M  - MAIN MENU
N  - NEXT ENTRY

ENTRY:
```

This menu, in combination with the output level attenuation setting on page 2-11, determines the final value of the output level. For example, if +7 dB is selected here, and the output level attenuation on page 2-11 is set to 5 dB, the resultant output level is a net gain of + 2 dB.

Regeneration Delay

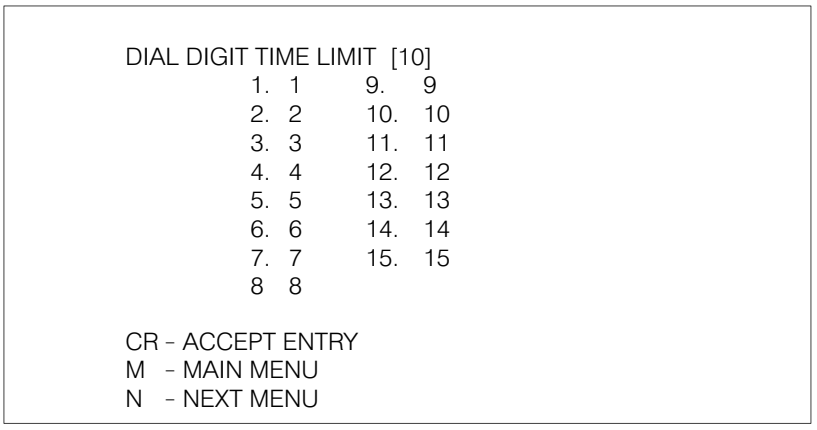
This menu is part of the Single Dial Tone function described on page 3-13. Ranging from 1 to 15 seconds, the selected delay compensates for the time it takes for the remote PBX to go off-hook and and be ready to accept dialing from the voice/fax channel for completing the call setup. Default is 1 second.

```
REGENERATION DELAY   [1]
        1.  1       9.   9
        2.  2       10.  10
        3.  3       11.  11
        4.  4       12.  12
        5.  5       13.  13
        6.  6       14.  14
        7.  7       15.  15
        8.  8

CR - ACCEPT ENTRY
M  - MAIN MENU
N  - NEXT MENU
```

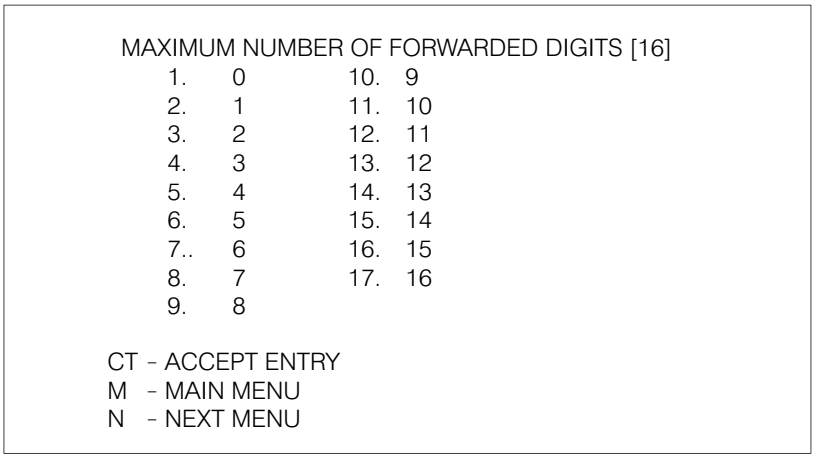
Dial Digit Time Limit

This option is part of the Single Dial Tone function described on page 3-13.



Ranging from 1 to 15 seconds, this option specifies the maximum time (from the start of one digit to the start of the next) for accepting dialed digits. When the time limit is reached, the accumulated digits are compared to the node's configured Number of Digits as defined on page 2-25. If the accumulated number is less than the configured Integration network number, a reorder (fast busy) response signal is generated. If the accumulated number is equal to or more than the configured number, a call request is generated. The default is 10 seconds. This allows a person time to pause and check a number being dialed. When using pulse dialing, this parameter must be set for 2 seconds or more to accommodate the high-order digits, such as 9 or 0 which take a longer time to record.

Maximum Number of Forwarded Digits



This menu is part of the Single Dial Tone feature described on page 3-13. Ranging from 1 to 16, the selected value specifies the maximum number of dialing digits to be forwarded to any destination voice channel in the network. If this number is reached, the call proceeds without having to wait until the Dial Digit Time Limit is reached.

The default value is 0, which will disable the Single Dial Tone feature. This assures backward compatibility, and requires positive action by the user to enable this feature.

When used in conjunction with a line configured to pulse dialing, this feature may be used to set limits on call routing. For example:

- A value of 4 or 5 will restrict calls to within a private network.
- A value of 8 will restrict calls to local calls from the PBX.
- A value of 9 will restrict calls to within the area code from the PBX.
- A value of 11 will restrict calls to within the country.
- Values beyond 11 would allow placing of international calls with resultant administrative call accounting issues.

Regeneration Format

```
REGENERATION FORMAT  [DIAL PULSE]
      1.  DIAL PULSE
      2.  DTMF

CR - ACCEPT ENTRY
M  - MAIN MENU
N  - NEXT MENU
```

This option is in support of the Single Dial Tone feature described on page 3-13. Designed for E&M and FXO interfaces, it allows the voice/fax channel to match the outgoing dialing digits with that of the interfacing PBX. Default is Dial Pulse. The voice/fax channel can accept either Dial Pulse or DTMF input digits.

Select DIAL PULSE for FXO and E&M interfaces if the interfacing PBX requires a dial pulse format. On FXS and FAX interfaces, selecting DIAL PULSE will result in no digits forwarded.

Select DTMF for FXO and E&M interfaces if the interfacing PBX requires a DTMF format. Select DTMF for FXS or FAX interfaces when used with PBXs or Key Systems equipped with DISA (Direct Inward System Access).

Call Progress Tones

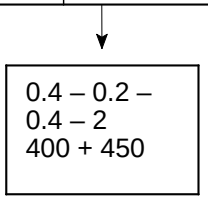
```
CALL PROGRESS TONES [N.AMER]
      1.  N. AMER
      2.  JAPAN
      3.  U.K.
      4.  EUROPE
      5.  FRANCE
      6.  C. AMERICA
      7.  CHILE
      8.  AUSTRALIA

CR - ACCEPT ENTRY
M  - MAIN MENU
N  - NEXT MENU
```

This menu allows you to match the call progress tones to one of eight countries or areas where the Integration network is installed. Details are listed in Table 2-6.

Table 2-6. Definition of Call Progress Tones

Option	Dial Cadence Frequency	Ringing Cadence Frequency	Busy Cadence Frequency	Congestion Cadence Frequency	Country
1	Continuous 350 + 440	2 – 4 440 + 480	0.5 – 0.5 480 + 620	0.25 – 0.25 480 + 620	North America
2	Continuous 440	1 – 2 384 + 416	0.5 – 0.5 400	0.5 – 0.5 400	Japan
3	Continuous 350 + 440	0.4 – 0.2 – 0.4 – 2 440 + 450	0.375 – 0.375 400	0.4 – 0.35 – 0.225 – 0.525 400	United King- dom
4	Continuous 425	1 – 4 425	0.5 – 0.5 425	0.25 – 0.25 425	Europe
5	Continuous 425 + 455	1.6 – 3.33 400	0.25 – 0.25 440	0.25 – 0.25 440	France
6	Continuous 425	1 – 4 425	0.25 – 0.25 425	0.25 – 0.25 425	Brazil, Mexico
7	Continuous 400	1 – 3 400	0.5 – 0.5 400	0.5 – 0.5 400	Chile
8	Continuous 400+425	0.4 – 0.2 – 0.4 – 2 400 + 450	0.375 – 0.375 425	0.375 – 0.375 425	Australia

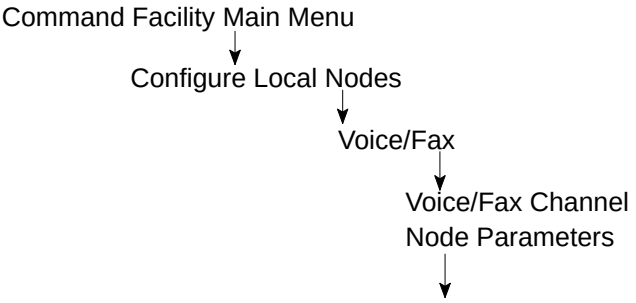


Example: In Australia, tones of 400 and 450 Hz are:

- on for 400 milliseconds
- silent for 200 milliseconds
- on for 400 milliseconds
- silent for 2 seconds

Setting the Voice/Fax Node Parameters

To configure the voice/fax channel node parameters, access the Command Facility Main Menu and follow this path:




```
VOICE/FAX NODE PARAMETERS
        1.  NUMBER OF DIGITS

CR - ACCEPT ENTRY
M  - MAIN MENU
P  - PREVIOUS MENU

ENTRY:
```

```
NUMBER OF DIGITS [2]
        1.  1
        2.  2
        3.  3
        4.  4

CR - ACCEPT ENTRY
M  - MAIN MENU
N  - NEXT ENTRY

ENTRY:
```

Ranging from 1 through 4, the value selected specifies the number of digits that need to be configured for all voice/fax extension numbers (channels) within the local node. Once configured, a voice/fax channel within any node can call only another voice/fax channel that has the same number of digits.

Configuring the Voice/Fax Switching Parameters

To configure the channel voice/fax switching parameters, access the Command Facility Main Menu and follow this path:

Command Facility Main Menu
↓
Configure Local Nodes
↓
Voice/Fax
↓
Prompt: **ENTER NODE-ID/CHANNEL # <^X TO ABORT>:**
↓
<cr>
↓
Typical Entry: **SIMI/E1** where Simi is the node ID, and E1 is voice/fax channel 1
↓
VOICE/FAX CHANNEL CONFIGURATION
↓
Voice/Fax Channel Configuration Menu
↓
VOICE/FAX SWITCHING PARAMETERS

```
VOICE/FAX SWITCHING PARAMETERS [node ID/channel#]
  1.  VOICE EXTENSION NUMBER                [XXXX]
  2.  FAX EXTENSION NUMBER                  [XXXX]
  3.  CALL INHIBIT                            [NO]
  4.  RECEIVE INHIBIT                         [NO]
  5.  AUTOCALL EXTENSION NUMBER             [XXXX]

CR - ACCEPT ENTRY
M  - MAIN MENU
P  - PREVIOUS MENU

ENTRY:
```

where xxxx shows the current number

Voice Extension Number

The VOICE EXTENSION NUMBER option of the Voice/Fax Switching Parameters Menu is used to assign the telephone extension number to the voice/fax channel. There is no default value. When this option is selected, the following display will appear on the screen:

```
ENTER VOICE EXTENSION (^X TO ABORT):
```

Enter 1 through 4 decimal digits. The number of digits entered must match the number specified in the voice/fax node parameters option (see page 2-25). If the entry does not contain the exact number of required digits, you will receive an error message.

To erase a prevously configured voice extension number, select VOICE EXTENSION option, then type in one space (space bar) followed by <cr>.

Fax Extension Number

This option is not supported by the Marathon 3K voice/fax channel.

Call Inhibit

YES, prevents the voice/fax channel from calling another voice/fax channel. Default is NO.

Receive Inhibit

YES, prevents the voice/fax channel from receiving calls. Default is NO.

Autocall Extension Number

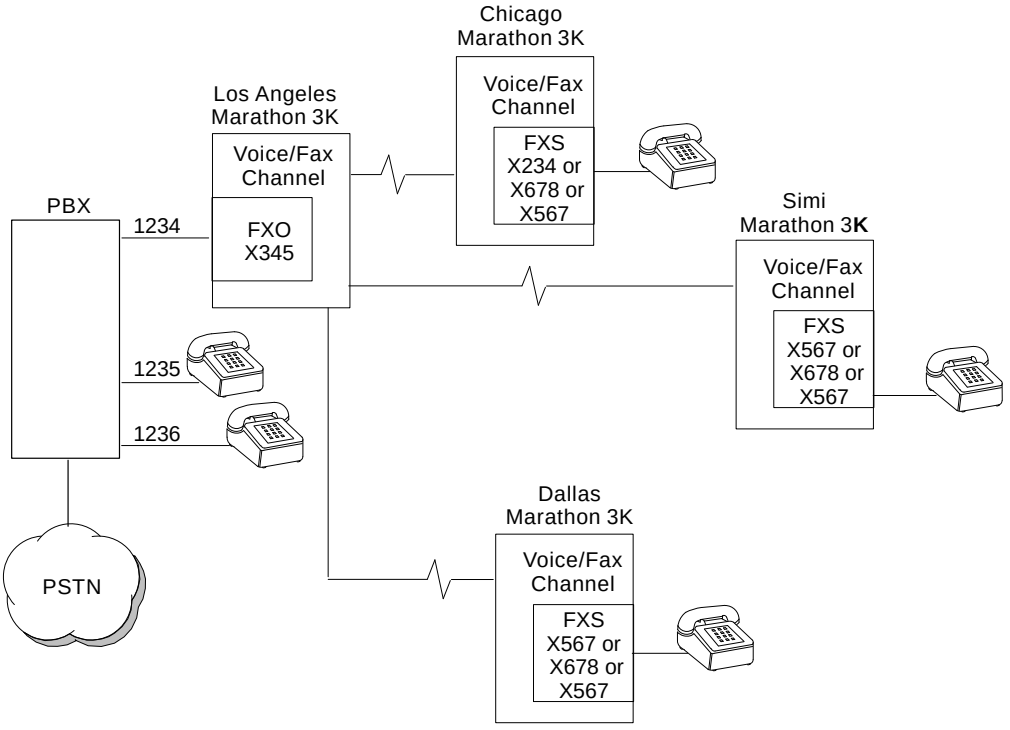
This option allows the local user to preselect a fixed destination (extension number) within the Integration network. Then, when the local voice/fax channel goes off-hook, it is automatically connected to that extension number.

Although outbound calls are directed to a single destination, the local voice/fax channel maybe accessed by multiple sources; when on-hook, it is free to receive calls from any of the other extensions within the Integration network provided that Receive Inhibit is not enabled.

Application

Autocall allows multiple users to contend for a limited number of resources within the Integration network. As shown in the following example, a headquarters office in Los Angeles is connected over three separate WAN links to field offices in Chicago, Simi, and Dallas. Each office is served by a single voice/fax channel. Specifically, the Los Angeles office is connected over an FXO-strapped voice/fax channel to PBX station 1234. The three field offices are strapped for FXS. Each voice/fax channel is assigned a 3-digit Integration network extension number as shown. In addition, the three field offices are configured with autocall enabled, and with autocall extension number configured for X345.

Any of the field office users can access the Los Angeles PBX by simply lifting the telephone off-hook and be automatically connected to PBX station 1234. From there the user can dial any other PBX station or dial an access code for a PSTN dial tone.



To Configure

To configure an autocall extension number, select AUTOCALL EXTENSION NUMBER from the Voice/Fax Switching Parameters Menu. The following display will appear on the screen:

```
ENTER AUTOCALL EXTENSION  (^X TO ABORT):
```

The destination number selected must match an already configured extension number within the Integration network.

To erase a prevously configured autocall extension number, select AUTO-CALL EXTENSION NUMBER, then type in one space (space bar) followed by <cr>.

Force Connecting a Voice/Fax Channel

As described earlier, voice/fax channels can be either switched or dedicated (force connected). Any channel that is not force connected is a switched channel. Normally, users connected to a switched voice/fax channel can call any other compatible voice/fax channel within the network by dialing the extension number of that channel.

In contrast, a force-connected voice/fax channel is paired up for connection with one other voice/fax channel only. It can neither call nor be called by other than the channel with which it is paired up, even when configured with an extension number. The procedure for force connecting a voice/fax channel follows.

Notes:

- The term “remote node” used in this procedure suggests that the other (mating) voice/fax channel is located in a remote Integration node, the most common application. You may, in fact, force connect any two voice/fax channels, including two within the same node or even within the same 3K Base Module. This is a convenient way to test voice/fax channels either before or after the network is in operation.
- Any voice/fax channel using AC15, Ron Tron, or wink start type signaling cannot be switched and must be force connected.
- Delay dial is not supported in force-connected channels

Force Connecting Mixed Channels

Do not force connect a voice/fax channel configured for G.729 with a voice/fax channel configured for any non-G.729 rate.

From the: Command Facility Main Menu
↓
Switching Control
↓
Force Connect
↓
Prompt: **ENTER FROM NODE-CHANNEL # (OR RANGE OF CHANNELS) (^X TO ABORT):**
↓
Typical Entry: SIMI/E1
↓
Prompt: **ENTER TO NODE-ID/CHANNEL # (^X TO ABORT):**
↓
Typical Entry: MPARK/E2

Entering this command will replace an existing connection with the new one.

Viewing a Status

If you wish to view the connection status of a voice/fax channel, select the Channel Status option from the Status/Statistics Menu. You will be prompted:

ENTER NODE-ID/CHANNEL# (OR RANGE OF CHANNELS) (^X TO ABORT)

Typical Entry

SIMI/E1

You will receive the following typical display:

[node ID] CHANNEL STATUS

CH	STATUS	SPEED	CONNECTED TO NODE/CHAN
E01	FCONN	8000	[MPARK , E02]
E02	OOS	8000	

ENTER CARRIAGE RETURN TO CONTINUE

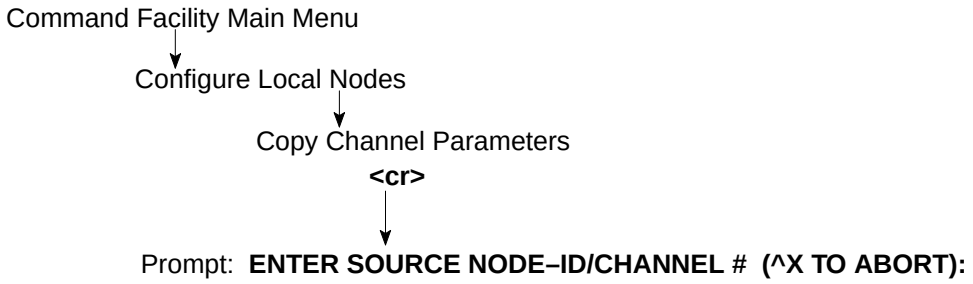
To disable or disconnect a force connection, select the FORCE DISCONNECT option in the Switching Control Menu. You will be prompted as follows:

ENTER FROM NODE CHANNEL # (OR RANGE OF CHANNELS) (^X TO ABORT):

Copy Channel Parameters

Having configured one voice/fax channel, you may copy the same parameters to another voice/fax channel (or a range of voice/fax channels). As a prerequisite, both voice/fax channels must be located within the same 3K Base Module, and must be strapped for the same interface type (FXO, FXS, E&M).

To enter the Copy Channel Parameters Menu, access the Command Facility Main Menu per page 2-6, and follow this path:


Command Facility Main Menu
Configure Local Nodes
Copy Channel Parameters
<cr>
Prompt: ENTER SOURCE NODE–ID/CHANNEL # (^X TO ABORT):

Enter the node id, a slash, and the channel number of the source voice/fax channel from where the parameters are to be copied, followed by a carriage return to accept the entry.

ENTER NODE-ID/CHANNEL (OR RANGE OF CHANNELS) (^X TO ABORT):

Enter the node id, a slash, and the channel number of the destination voice/fax channel where the configuration is to be copied, followed by a carriage return to accept the entry. The system will perform the copy and the screen will display the Command Facility Main Menu.

Important

If you copy a G.729-configured voice channel into a voice channel that does not incorporate the ClearVoice technology, you must reconfigure the digitizing rate of the non-G.729 channel. Otherwise, the non-G.729 channel will operate at the previous rate but will show a G.729 rate on the View Configuration display.

Network Code Download

Each voice/fax channel is equipped with a Flash memory that allows for local or remote updating of operating software.

When to Download

- To upgrade existing software.
- If the Voice/Fax Flash Status display indicates that an invalid or erased condition exists.

Refer to Appendix E for details.

Operation 3

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- E&M to FXS 3-7
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Single Dial Tone Operation 3-13
- Typical Application 3-14

This section provides step-by-step explanations of what happens when a call is placed in the various interface configurations. Single Dial Tone operation is also explained.

Voice/Fax Switching Operation

The following paragraphs describe how voice switching works in different telephone interface applications.

It is assumed that the voice/fax channel is enabled for switching operation (the default condition). If, for some reason, the channel comes up in a disabled state, you will hear a slow busy signal instead of a dial tone. To enable the channel for switching operation, return to the Command Facility Main Menu, and follow this path:

From the: Command Facility Main Menu
↓
Switching Control
↓
Enable Channel

ENTER NODE-ID/CHANNEL# (OR RANGE OF CHANNELS) ^X TO ABORT):

A typical entry would be:

SIMI/E1

where Simi is the node id, and E1 is voice/fax channel 1.

FXS to FXS

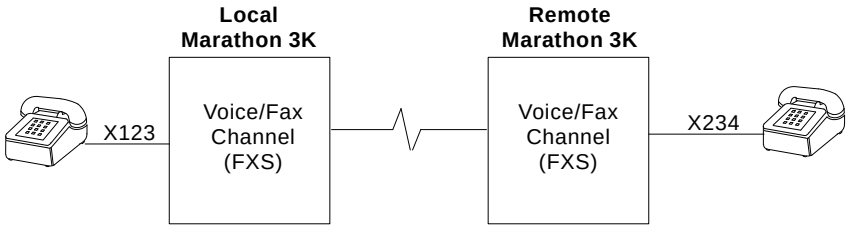

Figure 3-1. FXS-to-FXS Switching Operation

1. In Figure 3-1, the user at X 123 lifts the receiver. In response, the local voice/fax channel returns a dial tone.
2. The user at extension X 123 dials 234. The local Marathon 3K searches the network for an idle X 234. If X 234 exists but all such ports are busy, then a station busy (Slow Busy) tone will be sent to X 123. If X 234 cannot be found because it is not configured in the network, or the link to the node that contains X 234 is fully loaded or has failed, then a Reorder (Fast Busy) tone will be sent to X123.
3. Assuming that extension X 234 is not busy and is at the remote Marathon 3K, the remote unit returns a positive acknowledge signal.
4. The remote voice/fax channel rings X 234, and provides a ringback to the user at X 123.
5. Conversation starts when the user at X 234 lifts the receiver.
6. The connection is terminated when the user at either end hangs up the telephone.

FXS to FXO

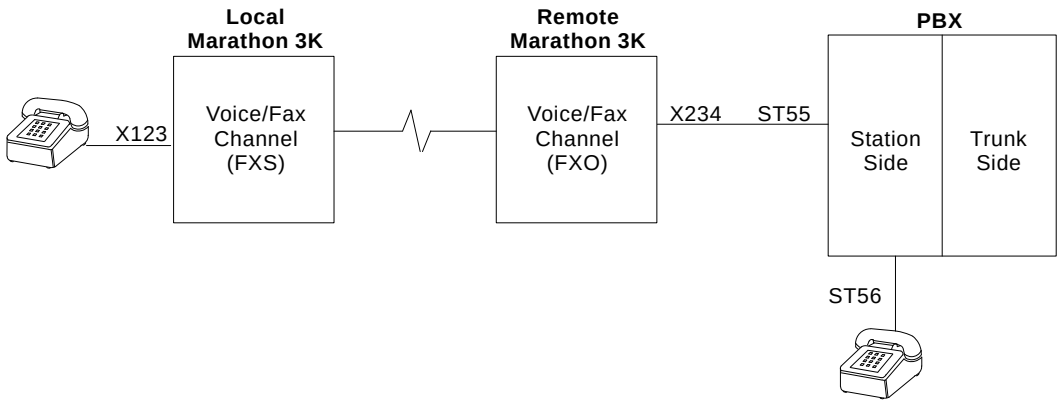

Figure 3-2. FXS-to-FXO Switching Operation

Note: In this application, there exists the possibility of a mismatch of impedances at the FXO interface. Voice quality and signalling may not be optimal.

1. In Figure 3-2, the user at X 123 lifts the receiver. In response, the local voice/fax channel returns a dial tone.
2. The user at X 123 dials X 234. In response, the local Marathon 3K first determines that X 234 is not a local extension. It then sends requests on all links to other Integration units within the system.
3. Assuming that extension X 234 is not busy, the remote Marathon 3K returns a positive acknowledge signal and connects the call to extension X 234. Since X 234 is connected directly to PBX station ST 55, the PBX returns a dial tone to the user at X 123.

Notes: If a call is coming in from extension 123 to station 55 while the PBX is in the middle of completing an outgoing call from station 55 to another extension, there is a slight chance that the user at extension 123 will be connected to the other extension.

To prevent this from happening, configure a few channels as outgoing only (receive inhibit) and others as incoming only (call inhibit). (See page 2-26.) To match these channels, the PBX port connecting to these channels would also have to be set as one-way, or assigned different extension numbers with Receive Only.

4. The user at X 123 dials ST 56, and conversation starts when the user at PBX ST 56 lifts the receiver. If ST 56 is busy, the PBX returns a busy signal to X 123.
5. The connection is terminated when the user at the FXS end hangs up the telephone. If the user at station 56 hangs up and the FXO can detect a valid disconnect from the PBX, then the network will terminate the connection.

FXO to FXS

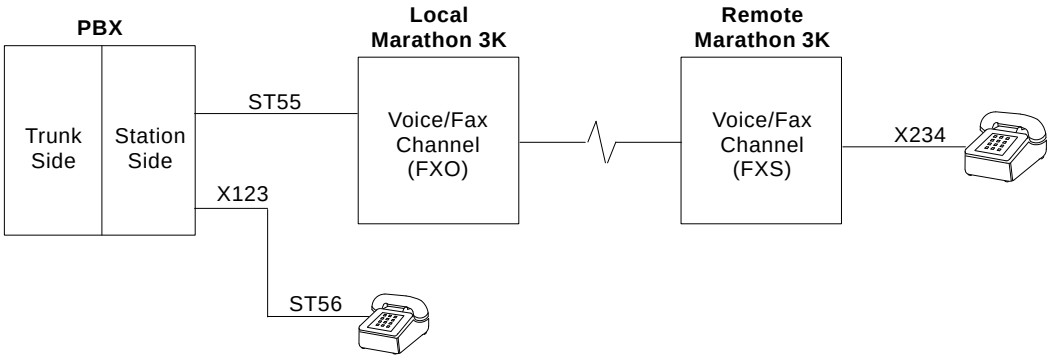

Figure 3-3. FXO-to-FXS Switching Operation

1. In Figure 3-3, the user at PBX station ST 56 lifts the receiver. Upon receipt of a dial tone from the PBX, the user dials PBX station ST 55. Following a preconfigured number of ringbacks (1 to 9) provided by the PBX, connection between ST 55 and X 123 is established, and the voice/fax channel within local Marathon 3K returns another dial tone to the user at ST 56.

   (See notes on page 3-4.)

2. The user at ST 56 dials X 234. In response, local Marathon 3K first determines that X 234 is not a local extension. It then sends requests on all links to other Integration units within the system.

3. Assuming that extension 234 is not busy, the remote Marathon 3K returns a positive acknowledge signal.

4. The voice/fax channel at the remote Marathon 3K rings extension X 234, and provides audible ringing tones to the user at extension X 123.

5. Conversation starts when the user at X 234 lifts the receiver.

6. The connection is terminated when the user at the FXS end hangs up the telephone. If the user at station 56 hangs up and the FXO can detect a valid disconnect from the PBX (power interrupt), then the network will terminate the connection.

E&M to E&M

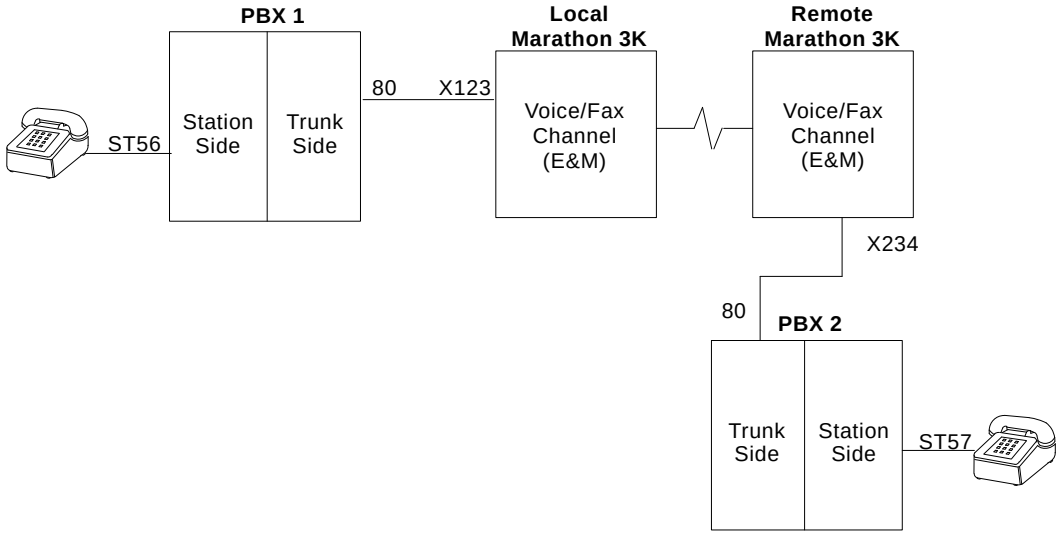

Figure 3-4. E&M-to-E&M Switching Operation

1. In Figure 3-4, the user at PBX 1, station ST 56, lifts the receiver. Upon receipt of a dial tone from PBX 1, the user dials access code 80, and is connected directly to the voice/fax channel in the local Marathon 3K.

2. Next, the user at ST 56 receives a dial tone from the voice/fax channel at the local Marathon 3K, and dials X 234.

3. The Local Marathon 3K first determines that X 234 is not a local extension. It then sends requests on all links to other Integration units within the system.

4. Assuming that extension X 234 is not busy, the remote Marathon 3K returns a positive acknowledge signal, and connects the call to extension X 234. Since X 234 is connected directly to tie line access code 80 on the trunk side of PBX 2, PBX 2 returns a dial tone to the user at ST 56.

5. The user at ST 56 dials PBX 2 station ST 57, and the PBX provides audible ringing tones. Conversation starts when the user at ST 57 lifts his receiver. If ST 57 is busy, PBX 2 returns a busy signal to ST 56.

6. The connection is terminated when the user at either end hangs up the telephone.

E&M to FXS

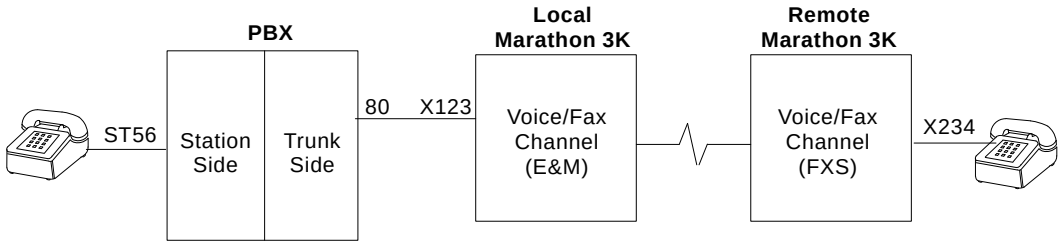

Figure 3-5. E&M to FXS Switching Operation

1. In Figure 3-5, the user at station ST 56 of the PBX lifts the receiver. Upon receipt of a dial tone from the PBX, the user dials access code 80, and is connected directly to the voice/fax channel in the local Marathon 3K.

2. Next, the user at ST 56 receives a dial tone from the voice/fax channel in the local Marathon 3K, and dials X 234.

3. The local Marathon 3K first determines that X 234 is not a local extension. It then sends requests on all links to other Integration units within the system.

4. Assuming that extension X 234 is not busy, the remote Marathon 3K returns an acknowledge signal.

5. The voice/fax channel at the remote Marathon 3K rings extension X 234, and provides audible ringing tones to the user at ST 56.

6. Conversation starts when the user at X 234 lifts the receiver.

7. The connection is terminated when the user at either end hangs up the telephone.

FXS to E&M

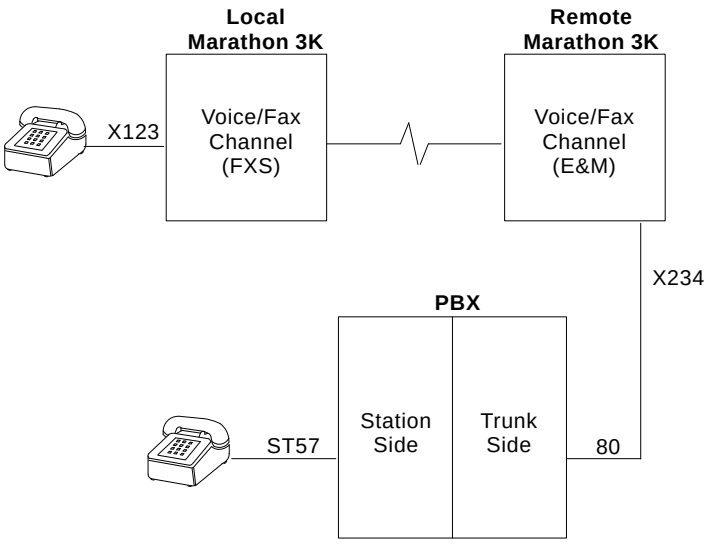

Figure 3-6. FXS to E&M Switching Operation

1. In Figure 3-6, the user at X 123 lifts the receiver. In response, the voice/fax channel returns a dial tone.

2. The user at X 123 dials 234. In response, the local Marathon 3K first determines that X 234 is not a local extension. It then sends requests on all links to other Integration units within the system.

3. Assuming that extension X 234 is not busy, the remote Marathon 3K returns a positive acknowledge signal, and connects the call to voice/fax extension X 234. Since X 234 is connected directly to the trunk side of the PBX, the PBX returns a dial tone to the user at X 123.

4. The user dials PBX station ST 57, and the PBX provides audible ringing tones. Conversation starts when the user at ST 57 lifts the receiver. If ST 57 is busy, the PBX returns a slow busy signal to X 123.

5. The connection is terminated when the user at either end hangs up the telephone.

FXO to FXO

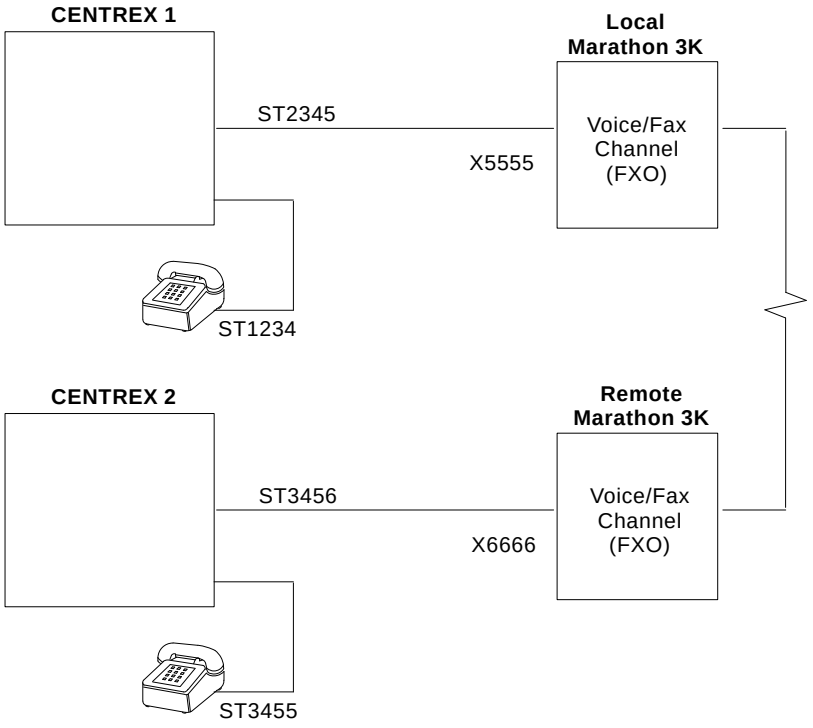

Figure 3-7. FXO to FXO Switching Operation

Note: In an FXO to FXO application, POWER INTERRUPT must be used in the Disconnect Supervision Menu (see page 2-18) of the FXO voice/fax channels at both end. Otherwise, the connection will not be established.

1. In Figure 3-7, the user at CENTREX 1, station 1234 lifts the receiver. Upon receipt of a dial tone from CENTREX 1, the user dials CENTREX 1, station 2345. Following a preconfigured number of ring cycles (1 to 9) provided by CENTREX 1, connection between CENTREX 1, station 1234 and the local voice/fax channel is established, and the local voice/fax channel returns another dial tone to the user at station 1234.

2. The user at CENTREX 1, station ST 1234 dials network extension X 6666. In response, local Marathon 3K first determines that X 6666 is not a local extension. It then sends requests on all links to other Integration units within the system.

3. Assuming that extension X 6666 is not busy, the remote Marathon 3K returns an acknowledge signal.

4. Next, the the remote voice/fax channel, which is always connected to CENTREX 2, station 3456, goes off-hook. In response, CENTREX 2 provides a dial tone to the user at CENTREX 1, station 1234.

5. The user at CENTREX 1, station 1234 dials station 3455, and CENTREX 2 provides audible ringing tones. Conversation starts when the user at station 3455 lifts the receiver. If station 3455 is busy, CENTREX 2 will provide a slow busy signal.

6. The connection is terminated when the user at either end hangs up the telephone.

FXO to E&M

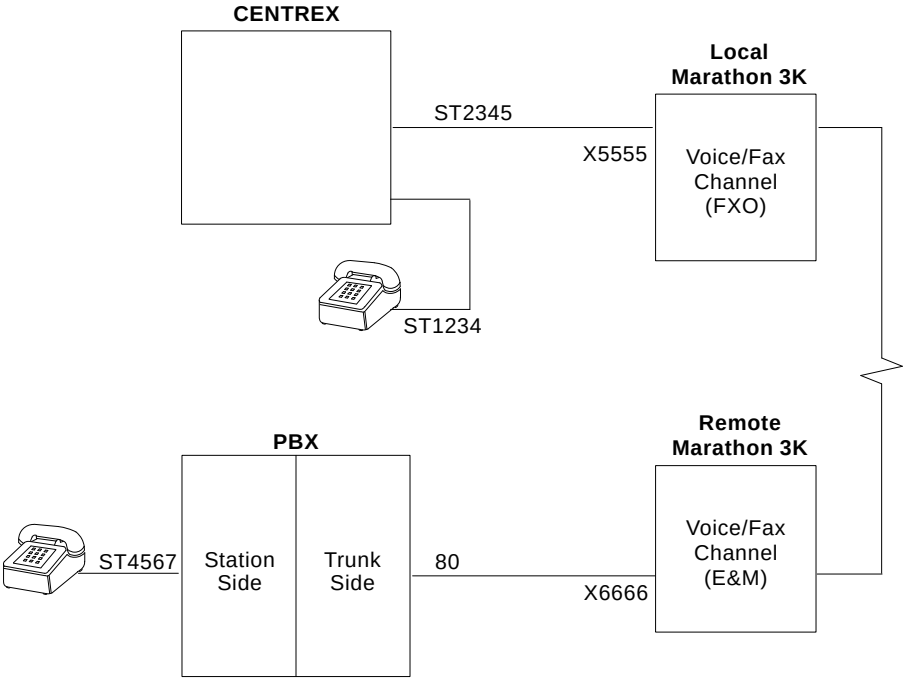

Figure 3-8. FXO to E&M Switching Operation

Note: For best results in an FXO to E&M application the POWER INTERRUPT option should be selected from the Disconnect Supervision Menu (see page 2-18) if the PBX can supply a power interrupt signal.

1. In Figure 3-8, the user at CENTREX station 1234 lifts the receiver. Upon receipt of a dial tone from the CENTREX, the user dials CENTREX station 2345. Following a preconfigured number of ring cycles (1 to 9) provided by the CENTREX, connection between CENTREX station 1234 and the local voice/fax channel established, and the local voice/fax channel returns another dial tone to the user at station 1234.

2. The user at CENTREX, station ST 1234 dials network extension X 6666. In response, the local Marathon 3K Integration unit first determines that X 6666 is not a local extension. It then sends requests on all links to other Integration units within the system.

3. Assuming that extension X 6666 is not busy, the remote Marathon 3K returns an acknowledge signal.

4. Next, the remote voice/fax channel, which is always connected to the trunk, goes off-hook. In response, the PBX provides a dial tone to the user at CENTREX station 1234. (Immediate dial trunk protocol.)

5. The user at CENTREX station 1234 dials PBX station 4567, and the PBX provides audible ringing tones. Conversation starts when the user at station 4567 lifts the receiver. If station 4567 is busy, the PBX will provide a slow busy signal.

6. The connection is terminated when the user at either end hangs up the telephone.

E&M to FXO

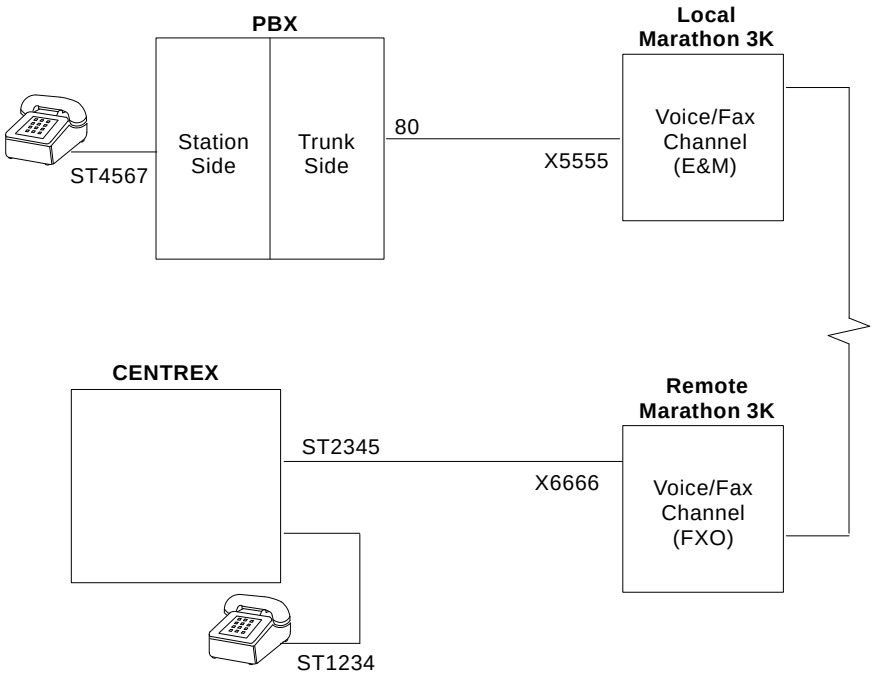

Figure 3-9. E&M to FXO Switching Operation

Note: For best results in an E&M to FXO application, select the POWER INTERRUPT option from the Disconnect Supervision Menu (see page 2-18) if the CENTREX can supply a power interrupt signal.

1. In Figure 3-9, the user at PBX station ST 4567 lifts the receiver. Upon receipt of a dial tone from the PBX, the user dials access code 80, and receives a dial tone from the local voice/fax channel.
2. The user at PBX station 4567 dials network extension X 6666. In response, the local Marathon 3K Integration unit first determines that X 6666 is not a local extension. It then sends request on all links to other Integration units within the system.
3. Assuming that extension X 6666 is not busy, the remote Integration unit returns an acknowledge signal.
4. Next, the remote Integration unit, which is always connected to CENTREX station 2345, goes off-hook. In response, the CENTREX provides a dial tone to the user at PBX station 4567.
5. The user at PBX station 4567 dials CENTREX station 1234, and the CENTREX will provide audible ringing tones. Conversation starts when the user at station 1234 lifts his receiver. If station 1234 is busy, the CENTREX will provide a slow busy signal.
6. The connection is terminated when the user at either end hangs up the telephone.

Slow Busy and Reorder

When a dialed extension is off hook, the caller will receive a slow busy signal. In most other circumstances an unsuccessful call attempt will result in a reorder (fast busy) response signal.

Reorder Response

The Integration Network switched telephone system will respond with a reorder (fast busy) signal when the call attempt was unsuccessful for one of the following reasons:

1. Called extension is force connected to a third telephone extension.
2. Called extension number does not exist.
3. When the link is overbooked, resulting in insufficient bandwidth to establish a call over the network.
4. One of the nodes within the network did not return a negative acknowledge message within a prescribed time limit, and all other polled nodes did return a negative acknowledge message.
5. Called extension is in the receive inhibit mode.
6. Caller did not enter the required number of digits within a prescribed time limit (1 to 15 seconds).
7. Lack of communication between the voice/fax channel and the Communications Control Module (CCM) on the 3K Base Module.

Special Notes

1. With the Single Dial Tone function disabled, calling a number with fewer digits than the number of digits configured (refer to page 2-25) is the same as calling a nonexistent number; it will result in a reorder (fast busy) signal.

2. With the Single Dial Tone feature enabled, calling a number with more digits than the number of digits configured, will result in a connection if the leading digits match an existing extension; otherwise it will result in a reorder (fast busy) signal.

3. If the link goes down while the channel is switch-connected to another channel, the call will be disconnected immediately, and a fast busy (reorder) will be generated. If the link recovers during the first 20 seconds of the timeout, connection will be restored. If the link does not recover in time, at the end of 30 seconds a reorder (fast busy) signal will be generated.

4. If the link goes down while the channel is force-connected to another channel, a slow busy signal will be generated. Connection will be re-established when the link is restored.

Single Dial Tone Operation

The main purpose of Single Dial Tone feature is to support Common Control Switching Arrangements (CCSA) on PBX networks through BLACK BOX's private Integration Network. In these arrangements, the user dials the number the same way as he does over the Public Switched Telephone Network. Then, the PBX must choose the most economical path for that call: BLACK BOX's private Integration Network, or the Public Switched Telephone Network. The PBX has the ability to delete the appropriate number of starting digits, and pass only that portion of the address required for the selected path.

To enable the Single Dial Tone feature requires that the maximum number of forwarded digits be configured for a non-zero value. Once this requirement is met, the remaining parameters of the Single Dial Tone feature are activated. These include the dial digit time limit, regeneration delay, and dial format.

With Single Dial Tone enabled, the Digital Voice Module accumulates dialing information beyond the 1 to 4 digits required for Integration Network access (see Voice Extension Number page 2-27). It continues to accept digits until either the maximum number of forwarded digits (1 to 16) is received, or until the configured dial digit time limit (the interval between the start of one digit and the start of the next digit) has been reached.

Typical Application

A typical application for the Single Dial Tone feature is shown in Figure 3-10. Here the Los Angeles PBX is connected to an Integration network using a Marathon 3K Integration node. It has links to three other locations: Dallas, Chicago, and Atlanta, each of which is equipped with a Marathon 3K Integration unit. The voice/fax channels at Los Angeles and Dallas are strapped for an E&M interface, and connected to the respective PBXs. The Atlanta Marathon 3K voice is configured for E&M interface with DTMF dial format and connected to the Central Office trunk of that PBX. The Chicago office is equipped with a Key Telephone system. Therefore the Marathon 3K at that location is configured for FXS interface with DIAL PULSE dial format.

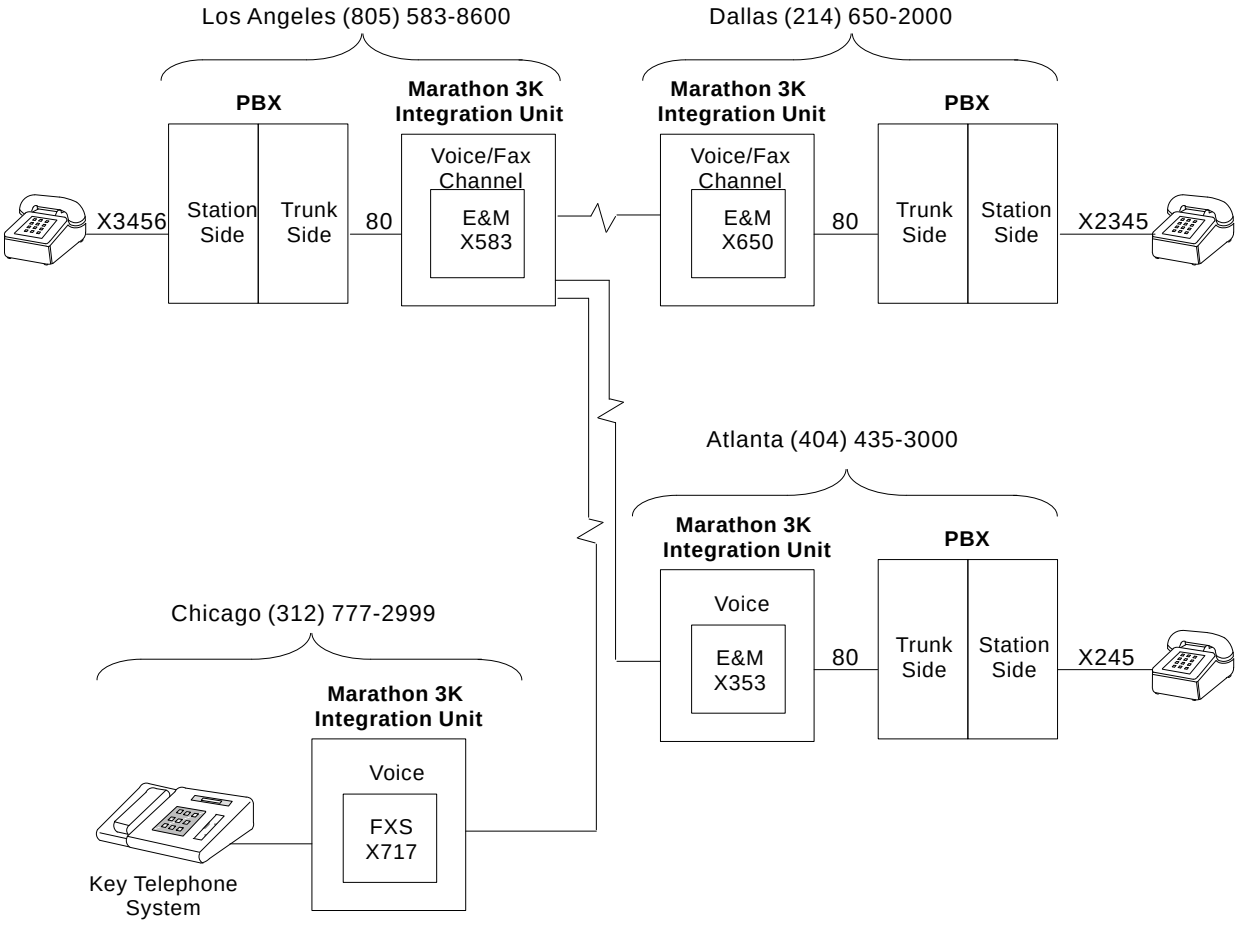

Figure 3-10. Typical Single Dial Tone Application

Assume that the user at the Los Angeles PBX station 3456 wants to call the users at the other company locations. The Integration Network is set for the three-digit dialing plan to best fit the prefix numbers of various offices (select the minimum number of digits to be unique beyond the forwarded digits – two digits would work in this example). Since there are four digits required at Dallas to identify the individual PBX stations, the voice/fax channel at Los Angeles will be configured for 4 Maximum Forwarded Digits. A Digit Time Limit of 1 second is selected, since the user would dial into the Los Angeles node in a continuous stream, without pause.

Calling Dallas

Without the Single Dial Tone feature enabled, the user at the Los Angeles PBX would have to dial 80, wait for a dial tone from the Marathon 3K, dial 650 to reach the Dallas voice/fax channel, wait for dial tone from the Dallas PBX, and then dial 2345 to reach the indicated individual in Dallas.

With Single Dial Tone enabled, the user at Los Angeles would dial 9 then 1(214)6502345. The PBX at Los Angeles would be programmed to look at the Area Code (214) and the Prefix (650) to determine the most economical routing. In this case, the PBX is programmed to first select Trunk Group 80 and see if there is any channel available on the Integration Network. Assuming that there is (otherwise the PBX would route the call over the PSTN), the PBX sends the last 7 digits (6502345) to the Los Angeles Marathon 3K. In turn, the Los Angeles unit uses the first 3 digits (650) to route the call to the Dallas unit immediately on receipt of the 7^{th} digit, and pass the last 4 digits (2345) to the Dallas unit. Here the voice/fax channel will wait 1 second (default regeneration delay) after seizing the Tie Trunk at Dallas, then send 2345 in either DTMF or Dial Pulse, as applicable, to the Dallas PBX. As a result, the Dallas PBX will either return a Busy signal if station 2345 is engaged, or ring station 2345 and return an audible ringing signal to the user in Los Angeles.

Calling Atlanta

Next, consider the user wanting to call station 245 in Atlanta. In this case, the user again calls the full telephone number 9 1(404)4353245. The PBX again selects trunk group 80 based on the Area Code (404) and the Prefix (435), but this time it is programmed to only send the last 6 digits (353245). The Los Angeles unit is expecting 7 digits, as before. Since there is no 7^{th} digit, the voice/fax channel will wait 1 second (the Dial Digit Time Limit) and then generate a call to Integration network address 353 and forward the last 3 digits (245). When the Atlanta unit receives the connect request, it will generate a ringing signal to the Central Office Trunk in the Atlanta PBX. This PBX trunk is configured for Direct Inward System Access (DISA) so that it automatically answers the call. The Atlanta voice/fax channel sends 245 as DTMF digits to the Atlanta PBX. The Atlanta PBX will then either send a busy tone if station 245 is engaged, or will ring station 245 and return an audible ringing signal to the user in the Los Angeles PBX.

Calling Chicago

The final case is a call to Chicago which is equipped with a Key Telephone System. The Los Angeles user again dials the full number for Chicago 9 1(312)7172999. The Los Angeles PBX again looks at the Area Code (312) and the Prefix (717), selects trunk group 80 and sends the last 7 digits to the Los Angeles unit. By sending the 7 digits, the Los Angeles unit will search for Integration Network address 717. The voice/fax channel on the Los Angeles unit will forward the last 4 digits to the unit at Chicago. In this case, the voice/fax channel is an FXS interface and set for Dial Pulse signaling. When this combination is encountered, the FXS interface in the Chicago unit node will discard any digits forwarded to it since the FXS interface cannot generate Dial Pulse signaling. As a result, the assigned line on the Key Telephone System will flash the associated lamp along with the ringing signal generated by the Marathon 3K.

In this environment, if the PBX in Los Angeles found all the channels associated with trunk group 80 busy, then any of these calls would have automatically been routed to the trunks which support the Inter Exchange Carrier that the customer in Los Angeles has chosen. Alternatively, there could be multiple layers of private trunk groups or multiple long distance carrier choices which the PBX would be programmed to choose, based on the least cost method of servicing each destination.

The above descriptions indicate the most complete use of the Single Dial Tone feature. If the PBX is unable to store and/or manipulate dialing digits, then the user in Los Angeles could call the defined trunk group 80 and access the Marathon 3K. At that point, the user could dial the complete Integration network number and extension as a single string. In this case, since the dialing is performed by a person, the Dial Digit Time Limit should be set in the range of 5 to 10 seconds to allow for pauses to check the next part of the number to dial.

Administration 4

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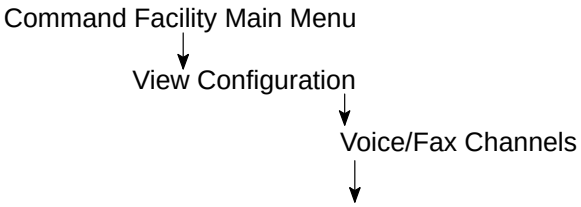
This section details the administrative functions of the voice/fax channel. Included are the viewing of configuration, status, statistics, hardware displays and messages.

View Configuration

The View Configuration feature allows you to view the current configuration values of any or all voice/fax channels installed within the local Marathon 3K unit, including the 3K Base Module. To view the current configuration, access the Command Facility Main Menu, and follow this path:



Enter the node ID/channel (or range of channels) followed by a carriage return <cr> to accept the entry and move to the screen display.

There are four screen displays associated with the voice/fax channels. Three of the displays deal with channel characteristics, while the fourth display shows the configured switching parameters. Following is a sample of the three voice/fax screen displays. Each item in the display is explained on page 4-5. To advance through the displays, press the <cr> key. Pressing the <cr> key after the fourth display will return the program to the View Configuration Menu.

Notes:
- Only the voice/fax channels incorporated within the 3K Base Module are included in the sample.
- The logical location of the voice/fax channels in the Marathon 3K unit is module location E. Accordingly, channel 1 is displayed as E01 and channel 2 is displayed as E02.

Screen 1

```
[node ID] VOICE/FAX CHANNEL CHARACTERISTICS
                OPERATION                INPUT   OUTPUT   BUSYOUT
CH.   RATE      ANALOG SIGNALLING        GAIN    ATTN     MODE            BACKGROUND
-------------------------------------------------------------------------------------
E01   G729      2-WIRE  FXS INTERRUPT    0 dB    0 dB     SYSTEM CNTRL    REGENERATED
E02   4800      2-WIRE  FXO              0 dB    0 dB     SYSTEM CNTRL    SILENCE

ENTER CARRIAGE RETURN TO CONTINUE
```

Screen 2

```
VOICE/FAX CHANNEL CHARACTERISTICS
      RINGING                                                          NUMBER
CH.   FREQUENCY   MODE          BANDWIDTH    IMPEDANCE   PRIORITY      OF RINGS   AGC
-------------------------------------------------------------------------------------
E01   25 HZ       VOICE/FAX     VOICE ACT    COMPLEX     HIGH          --         --
E02   --         VOICE/FAX     CONTINUOUS   600 OHMS    HIGH          3

ENTER CARRIAGE RETURN TO CONTINUE
```

Screen 3

```
VOICE/FAX CHANNEL CHARACTERISTICS
        FAX   DISCONN          MAX OUT   DIAL  REGEN                 REGEN  FWD     COMPDR
CH.     RATE  SUPERVSN         LEVEL     LIMIT FORMAT   CPT         DELAY  DIGITS  FORMAT
-------------------------------------------------------------------------------------------
E01     9600  - -              - -       10    PULSED   N.AMER      1      16      - -
E02     9600  POWER INTER      - -       10    DTMF     N.AMER      1      16      - -

ENTER CARRIAGE RETURN TO CONTINUE
```

Screen 4

```
VOICE/FAX SWITCHING PARAMETERS          NUMBER OF DIGITS: [4]

        VOICE          FAX                                              AUTOCALL
CH.     EXTENSION      EXTENSION      CALL INH   RCV INH    SWITCH      EXTENSION
-------------------------------------------------------------------------------------
E01     1234           - -            NO         NO         ENABLE      - -
E02     1235           - -            NO         NO         ENABLE      - -

ENTER CARRIAGE RETURN TO CONTINUE
```

Following is a definition of the items displayed in the voice/fax portion of the View Configuration display:

Ch – The physical location of the voice/fax channel in the local Marathon 3K unit. The channel designations for the voice/fax portion of the 3K Base Module are always E01 and E02.

Rate – The voice digitizing rate. Values range from 4000 bps to 16000 bps and G.729 (8 Kbps).

Operation – The way the voice/fax channel is strapped and configured.

Analog – E&M-strapped voice/fax channels may be strapped for either 2-wire or 4-wire operation. All FXS- and FXO-strapped voice/fax channels are fixed for 2-wire operation.

Signaling – Interface type and, if E&M, type of signaling convention used. May include the following:

FXS Interrupted Ring
FXS Repeated Ring
E&M DC
E&M 2280
E&M Pulsed DC

Input Gain – The setting of the input level gain. Values range from –6 dB to 19 dB.

Output Attenuation – The setting of the output level attenuation. Values range from 0 dB to 25 dB.

Busyout Mode – A method for setting the voice/fax channel in the busy state.

Background – Method for managing background noise (silence or regenerated).

Ringing Frequency – Either 25 Hz or 50 Hz ringing frequency selected for the FXS voice/fax channel.

Mode – Voice/fax or voice only.

Bandwidth – Method for utilizing link bandwidth for voice/fax channel.

Impedance – Value selected to match the line impedance of the telephone equipment interfacing with the voice/fax channel.

Priority – When high, voice/fax channel has priority over synchronous data channel when contending for link bandwidth.

Number of Rings – The number of ringbacks returned by the FXO voice/fax channel before providing a dial tone.

AGC – (Automatic Gain Control) not implemented.

Fax Rate – The digitizing rate selected for fax operation. See page 2-17.

Disconnect Supervision – Type of signal (tone or power interrupt pulse) generated by the PBX to indicate that a station user hung up the telephone. The FXO voice/fax channel uses this information to break the voice/fax connection over the Integration network.

Maximum Output Level – Special setting for a voice/fax channel strapped for E&M and configured for 4-wire operation. Choices are 0 dBm and +7 dBm.

Dial Limit – Configured interdigit time limit. Timer is reset after each digit received. If the next digit is not received before time expires, the voice/fax channel will stop accumulating further digits.

REGEN Format – Pulse or DTMF, as configured, to match that of the associated PBX.

CPT (Call Progress Tone) – Set to match the calling convention tones of the country or region of operation.

REGEN Delay – Allows for time delay while the PBX prepares to receive regeneration dialing digits.

FWD Digits – Maximum number of received digits to be forwarded. When that number is reached, the digits are forwarded without waiting for the time limit to expire.

COMPDR Format – Not supported in the 3K voice/fax facility.

Voice Extension – The voice extension number presently configured for that voice/fax channel.

Fax Extension – Not supported in the 3K voice/fax facility.

Call Inhibit – Indicates whether the voice/fax channel can or cannot call another voice/fax channel within the Integration network.

Receive Inhibit – Indicates whether the voice/fax channel can or cannot receive calls from another voice/fax channel within the Integration network.

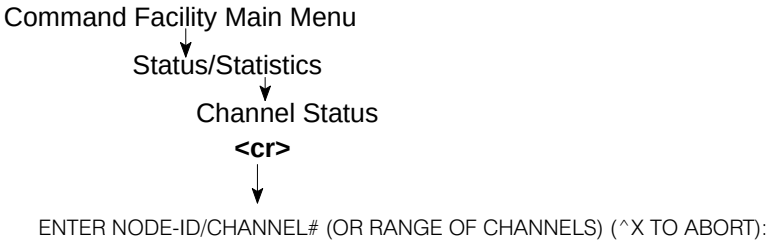
Switch – Indicates whether or not the voice/fax facility is configured for switching.

Autocall Extension – The extension number configured to be dialed automatically when going off-hook.

Status Displays

Note: Only the voice/fax channels incorporated in the 3K Base Module, E01 and E02, are treated here.

There are two status displays relating to the voice/fax channel: channel status, and voice/fax status. To obtain these status reports, access the Command Facility Main Menu and follow this path:



Enter the node ID and channel number (or range of channels) followed by a carriage return. You will obtain the following typical display:

```
[node ID] CHANNEL STATUS

                                        CONNECTED
CH          STATUS     SPEED            TO NODE/CHAN
-------------------------------------------------------

E01         FCONN      G.729            [MPARK   , B02]
E02         OOS        8000
```

IDLE – Indicates that the channel is available but currently not used. Not shown in the above sample.

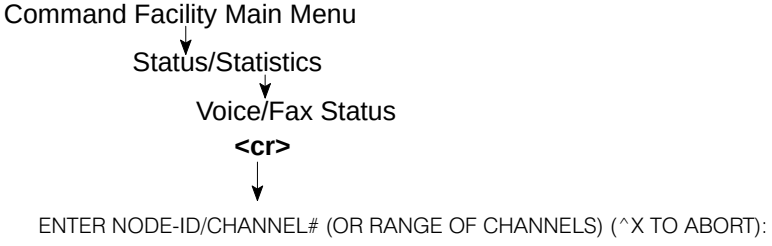
CONN – Indicates that the channel is switch connected. Not shown in the above sample.

FCONN – Indicates that the channel is force-connected to another channel, as shown in the "CONNECTED TO" column.

OOS – Indicates that the channel has been disabled or has failed.

CIP – Connection in process. Not shown in the above sample.

Next, reenter the Status/Statistics Menu from the Command Facility Main menu, and follow the path:



Enter the node ID and channel number (or range of channels) followed by a carriage return. You will obtain the following typical display:

```
[node ID/ch#] VOICE/FAX STATUS   TIME hh:mm:ss   DATE mm/dd/yy
                 INPUT                  SOFTWARE   TEST                    TEST
CH.   RATE       LEVEL      STATE       REVISION   MODE                    STATUS
-------------------------------------------------------------------------------------------
E01   G.729      <–25 DBM   ON HOOK     2249–00A    – –
E02   8000       <–25 DBM   BUSY        2249–00A   LOCAL SELF TEST         PASS

ENTER CARRIAGE RETURN TO CONTINUE

                                CHANNEL   INTERFACE          INTERFACE       FLASH
CH.   SWITCH  MODE              RECOVERY  NUMBER     REV     DESCRIPTION     STATUS
-------------------------------------------------------------------------------------------
E01   ENABLE   VOICE             00        1234       C      ENHANCED FXS    ACTIVE
E02   ENABLE   FAX               00        5678       E      ENHANCED FXO    ERASED
```

Ch. – The physical location of the voice/fax channel in the Marathon 3K unit. The channel designation for the voice/fax portion of the 3K Base Module are always E01 and E02.

Rate – The rate in bps at which the voice channel is currently operating. If a channel is configured with DYNAMIC bandwidth, the value will fluctuate. If a channel is configured for fixed bandwidth, the value will be the configured rate.

If a channel operates in the fax mode, the displayed value indicates the rate at which the channel transmits (not receives) fax messages.

Input Level – The value of the input signal monitored, in dBm.

State – State of current operation. ON HOOK, and OFF HOOK apply to force-connected channels, while IDLE, CONNECT and BUSY apply to switched channels.

Software # Revision – Type and revision of the voice/fax software.

Test Mode – Type of test, if any, currently running.

Test Status – The status of the current test, if in the test mode.

Switch – Whether or not switching is enabled.

Mode – The mode (voice/fax or voice only) the voice/fax channel is operating in.

Channel Recovery – is a counter value, ranging from 00 through 99, that indicates how many times the channel was reset by software. This counter value will return to zero when the system is reinitialized.

Interface Number – Not applicable to the voice/fax channel incorporated in the 3K Base Module.

Interface Revision – Not applicable to the voice/fax channel incorporated in the 3K Base Module.

Interface Description – Describes the voice/fax channel's type of telephone interface (FXS, FXO, to E&M).

Flash Status – Status of the channel's Flash EPROM, coded as follows:

Not Ins = Not installed.

Active = Loaded and operational.

Erased = Ready to receive file.

Invalid = Checksum Error or other fault.

BAD RCV = File was incomplete due to transmission error or problem in receiving. Reset voice/fax channel, erase Flash and try again.

Note: *Channel Inoperable* will be displayed in the Voice/Fax Status Menu if the hardware is defective.

Statistics Display

The Marathon 3K maintains a running inventory of statistical information on voice/fax channel activity. Periodically, or upon special request, the accumulated statistical information is displayed on the screen for a particular node.

Statistical information monitored includes channel number, duration and type of calls made, calls attempted, duration of busy condition, number of times voice/fax frames were discarded due to channel congestion, faxing time, and percentage of calls in fax mode.

There are four different functions that display voice/fax channel statistics display, as follows:

- Periodic Report
- System Statistics
  - Last Period Report
  - Demand Report
- Channel Statistics
- Daily Voice/Fax Statistics

Periodic Display

Voice statistics forms part of the periodic statistics report, and is appended to the channel portion of the display. Frequency of display is a value selectable in intervals of

- None
- 10 minutes
- 20 minutes
- 30 minutes
- 60 minutes

When **none** is selected, the information is cleared on the hour but not displayed automatically; only on special request (on demand).

The voice/fax channel portion of the periodic report has the following format:

VOICE/FAX STATISTICS

CH.	TOTAL TIME	TOTAL CALLS	AVERAGE DURATION	CALL ATTEMPTS	BUSY OUT	CHANNEL CONGESTION	FAX TIME	FAX%
E01	25:14	7	03:30	7	00	0	05:00	20
E02	12:30	4	03:00	4	00	0	03:10	25

Following is a definition of the items displayed in the voice/fax portion of the periodic display:

Ch. – The physical location of the voice/fax channel in the Marathon 3K unit. The channel designations for voice/fax portion of the 3K Base Module are always E01 and E02.

Total Time – Summation of time when the local voice/fax channel was connected to any remote voice/fax channel. Time given in minutes and seconds.

Total Calls – Number of completed connections during interval, whether initiated locally or from the remote end. Values range from 00 through 99.

Average Duration – Average time of all calls detected during the periodic interval. Equals total time divided by number of incoming and outgoing calls (in minutes and seconds). All averages are computed at the end of each call.

Call Attempts – Number of times the local voice/fax channel went off-hook. Values range from 00 through 99.

Busyout – Length of time the local voice/fax channel has lost synchronization with the remote channel (in minutes and seconds).

Channel Congestion – Number of times voice frames have been discarded during the reporting period, with values ranging from 0 through 999; *** indicates more than 999. High counts or *** define congestion at the link level that require reallocation of system channel bandwidth. The system administrator can do this by increasing the composite link bandwidth or reducing the channel rates either directly or by using the DYNAMIC bandwidth option. To select the DYNAMIC bandwidth option, refer to page 2-12.

Fax Time – Amount of time in minutes and seconds that the voice/fax channel has been operating in the fax mode.

Fax% – Percentage of time the voice/fax channel was operating in the fax mode.

System Statistics

System statistics may be obtained in two ways: through the Last Period Report or Demand Report. To enter the System Statistics Menu, you must first select Status/Statistics from the Command Facility Main Menu, followed by System Statistics. The display is the same as shown on page 4-10.

Last Period Report

When Last Period Report is selected, the screen will display the same system-related statistics information, including voice/fax statistics, that was provided at the last report.

Demand Report

When Demand Report is selected, the screen will display system-related statistical information, including voice/fax channel statistics, accumulated since the last periodic report.

Channel Statistics

Channel Statistics is another statistics option. It deals only with channel-related information. When selected, the screen will display channel-related statistical information, including voice/fax channel statistics, accumulated since the last periodic report.

To enter the Channel Statistics Menu, you must first select Status/Statistics from the Command Facility Main Menu, followed by Channel Statistics. The display is of the same format as the periodic report display described on page 4-10.

Voice/Fax Daily Statistics

Daily Voice/Fax Statistics is a display of up-to-the-minute voice/fax data accumulated since midnight. At midnight all statistics of the previous 24 hours are cleared and a new 24-hour cycle started. To obtain a complete periodic report, it should be printed just before midnight.

To enter the Voice/Fax Statistics Menu, use the following path:

Command Facility Main Menu
↓
Status/Statistics

The information is displayed in the following format:

[node ID/ch#] VOICE/FAX STATISTICS TIME hh:mm:ss DATE mm/dd/yy

CH.	TOTAL TIME HRS:MIN	TOTAL CALLS	BUSYOUT HRS:MIN	FAX TIME HRS:MIN	FAX%
E01	06:25	69	00:02	00:30	08
E02	02:30	22	00:00	00:20	13

ENTER CARRIAGE RETURN TO CONTINUE

Ch. – The physical location of the voice/fax channel in the Marathon3K unit. The channel designations for voice/fax portion of the 3K Base Module are always E01 and E02.

Total Time – Summation of time when the local voice/fax channel was connected to any remote voice/fax channel. Time given in hours and minutes.

Total Calls – Number of completed connections during interval, whether initiated locally or from the remote end. Values range from 00 through 99.

Busyout – Length of time the local voice/fax channel has lost synchronization with the remote channel (in hours and minutes).

Fax Time – Amount of time in hours and minutes that the voice/fax channel was operating in the fax mode.

Fax% – Percentage of time the voice/fax channel was operating in the fax mode.

Hardware Display

This option displays the type of hardware currently installed in each of the Integration unit module locations.

To obtain the hardware display, follow this path:

Command Facility Main Menu
↓
View Configuration
↓
Hardware

The information is displayed in the following format (only the voice/fax portion is displayed here).

```
[node ID]  HARDWARE CONFIGURATION
SLOT        MODEL NUMBER          REVISION     MODULE DESCRIPTION
-------------------------------------------------------------------------------
E                                 – –          2 CH INTEGRAL VOICE
D           SLOT EMPTY            – –          – –
C           SLOT EMPTY            – –          – –
B           SLOT EMPTY            – –          – –
A           100–3606–002          G            3K CCM

ENTER CARRIAGE RETURN TO CONTINUE
```

Messages

Messages related to the voice/fax channel appear on the Command Facility terminal and are listed in Table 4-1. The terms used in the Message Type column are defined below:

Event — Something which occurs in the system during regular operations; most likely will not affect the system's operation.

Diag — Diagnostic messages

Info — Informational. These messages provide the system administrator with knowledge of various situations.

Table 4-1. Screen Display Messages

Message	Message Type	Explanation
CHANNEL INOPERABLE	Info	You have requested status for a voice/fax channel that is not operational.
CHANNEL NOT INSTALLED	Info	An attempt has been made to configure or test a voice/fax channel. However, the channel number selected is not a voice/fax channel.
CHANNEL(S) NOT INSTALLED	Info	An attempt has been made to view the status or configuration of channels that are not voice/fax channels.
CHANNEL NOT OPERATIONAL	Info	A voice/fax channel you are trying to test is not working properly.
ILLEGAL INPUT	Info	The number of digits entered (voice extension number) is less or more than the number of digits configured for extension numbers.
ILLEGAL SELECTION	Info	You have entered a number that is greater than the number of options shown. (For example, the available options are 1, 2, 3, and 4, and you have entered the number 5 or greater). Try again.
INVALID NUMBER	Info	1. The number entered is less or more than the number of digits configured for extension numbers. 2. The input requested by the prompt requires a numeric response, and there was no numeric response.
NOT SUPPORTED	Info	The requested voice/fax configuration parameter selected is not supported by the voice/fax channel. For example, a fax extension number.
NUMERIC REQUIRED	Info	You must enter a numeric character. Try again.
OUT OF RANGE	Info	The range of numbers entered is not available in this unit.

Table 4-1. Screen Display Messages (continued)

Message	Message Type	Explanation
VOICE CHANNEL [*n/c*] <condition> BUSY OUT	Event	Voice/fax channel *c* on node *n* has been put in a busyout condition. *Conditions are the following:* **Force Connect** 1. System. The link has experienced a sync loss, a force disable command, or a force disconnect command. 2. Force On. The network operator has put the channel in a busyout condition via a Forced On busyout command. **Switched** The network operator has put the channel in busyout condition via a Forced On busyout command.
VOICE CHANNEL [*n/c*] BUSYOUT CLEARED	Event	Termination of a busyout condition has occurred on voice/fax channel *c*.
VOICE CHANNEL [n/c] INPUT LEVEL: *nn* dBm	Event	The current dBm level for voice/fax channel *c* on node *n* is nn (-25 to +1) dBm.
VOICE CHANNEL *[n/c]* LOCAL LOOP TEST	Diag	Voice/fax channel *c* on node *n* has started the diagnostic local loop test.
VOICE CHANNEL [*n/c*] RESET	Event	The voice/fax channel *c* on node *n* has been reset and re-initialized with the current configuration.
VOICE CHANNEL *[n/c]* SELF TEST FAIL	Diag	Voice/fax channel *c* on node *n* has completed the internal diagnostic tests unsuccessfully.
VOICE CHANNEL [*n/c*] SELF TEST PASS	Diag	Voice/fax channel *c* on node *n* has completed the self diagnostic tests successfully.
VOICE CHANNEL *[n/c]* TEST CLEARED	Event	Voice/fax channel *c* on node *n* has cleared (stopped) the current diagnostic test.

Legend:
n/c Indicates node name (n) and channel number (c).

Reset Voice/Fax Channel

The RESET VOICE CHANNEL option of the Voice/Fax Tests Menu is a means for resetting each voice/fax channel individually without affecting the operation of any other channel within the Marathon 3K; it clears any switched voice/fax channel operation currently in progress, but does not clear a force connect. The same function is available as a channel reset in the reset menu.

Testing 5

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- Testing of Two Local Voice/Fax Channels 5-17

This section describes how to test the voice/fax portion of the 3K Base Module. All functions can be effected from the Command Facility Main Menu, using an asynchronous terminal connected to any one of the asynchronous ports.

Also, you may connect the two channels together and test them for both voice quality and switching capability.

To test the voice/fax channel enter the Command Facility Main Menu and follow this path:

Command Facility Main Menu
↓
Diagnostics
↓
Prompt: **ENTER NODE-ID/CHANNEL #(OR RANGE OF CHANNELS)(^X TO ABORT):**

A typical entry would be:

SIMI/E1
↓
where Simi is the node ID, and E1 is voice/fax channel 1

Voice/Fax Tests

```
VOICE/FAX TESTS
        1. SELF TEST
        2. LOOPBACK
        3. INPUT LEVEL DISPLAY
        4. RESET VOICE/FAX CHANNEL
        5. TERMINATE TEST

CR  - ACCEPT ENTRY
M   - MAIN MENU
ENTRY:
```

There are basically two tests that you can perform from this menu: loopback and self test. The other options in this menu are additional diagnostic tools and test controls.

A typical sequence of tests is given in the following paragraphs. You may find a different sequence more suitable to your needs.

Self-Test

You may perform this test any time after power has been applied to the Marathon 3K unit. If successful, the test verifies that the voice/fax channel can convert analog signals into digital and back to analog. The procedure is as follows:

```
VOICE/FAX TESTS
      1. SELF TEST
      2. LOOPBACK
      3. INPUT LEVEL DISPLAY
      4. RESET VOICE/FAX CHANNEL
      5. TERMINATE TEST
```

After selecting the self test option, observe the green OK indicator for the applicable voice/fax channel on the front of the 3K Base Module. If the OK indicator is on, the test was successful. If the OK indicator is off or flickering, the test failed.

Note: The results of the self-test will also be shown in the display on the Command Facility terminal screen.

If the test was successful, the display will show:

```
VOICE/FAX CHANNEL [SIMI/E1] SELF TEST PASS
```

If the test failed, the display will show:

```
VOICE/FAX CHANNEL [SIMI/E1] SELF TEST FAIL
```

Stop the self-test by selecting TERMINATE TEST, followed by a carriage return. The display will show:

```
VOICE/FAX CHANNEL [SIMI/E1] TEST CLEARED
```

Test Connections

Connections for all switched and force-connected loopback tests are made by modular cable at the rear of the 3K Base Module, as shown in Figure 5-1. The modular cables for FXO and FXS connections are supplied. The modular cable for E&M connections is available as an option.

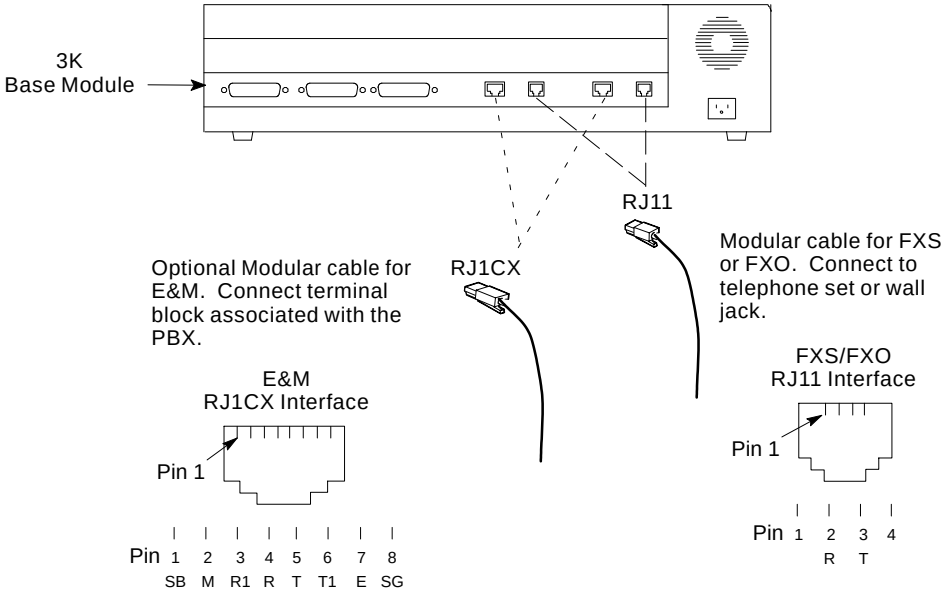

3K
Base Module
RJ11
Modular cable for FXS or FXO. Connect to telephone set or wall jack.
Optional Modular cable for E&M. Connect terminal block associated with the PBX.
RJ1CX
E&M
RJ1CX Interface
Pin 1
Pin 1 2 3 4 5 6 7 8
SB M R1 R T T1 E SG
FXS/FXO
RJ11 Interface
Pin 1
Pin 1 2 3 4
R T

Figure 5-1. Loopback Connections

Loopback Tests, Switching Mode

```
VOICE/FAX TESTS
      1. SELF TEST
      2. LOOPBACK
      3. INPUT LEVEL DISPLAY
      4. RESET VOICE/FAX CHANNEL
      5. TERMINATE TEST
```

When the voice/fax channel is in the switching mode, the loopback test is single ended. If successful, it verifies that all circuits of the voice/fax channel, including the telephone interface circuits, are working properly. In the following examples all extension numbers are fictitious; the actual extension numbers, if any, are determined by your installation.

When the voice/fax channel is force connected to a remote voice/fax channel, both ends are in loopback, and the TM indicator should go on at both ends.

FXS Telephone Interface

After selecting the LOOPBACK option of the Voice Fax Tests Menu, observe the indicators of the voice/fax channel being tested. The indicators should assume the following states:

OK	LO	RO	TM
0	1	0	1

Legend:
where 0 = OFF
1 = ON

1. See the test hookup shown in Figure 5-2. Use the modular cable supplied. Connect cable as shown in Figure 5-1.
2. Lift the telephone receiver. The channel's LO indicator should go on and stay on.
3. Speak into the microphone. You should hear your own voice echoed back to you. It is normal to hear more than one echo of your voice.

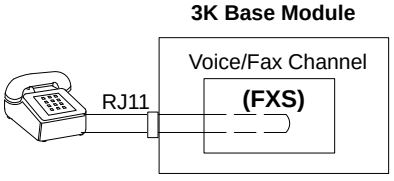


Figure 5-2. Example of a Switching FXS Loopback Test

Note: If you hear a busy tone, place the voice/fax channel temporarily into the FORCED OFF busyout mode (refer to page 2-11). Be sure to restore this mode to SYSTEM CONTROLLED at the end of this test.

4. Hang up the telephone.
5. Stop the loopback test. To do that, select the TERMINATE TEST option. Verify that the LO and TM indicators on the channel being tested are off.

VOICE/FAX TESTS
1. SELF TEST
2. LOOPBACK
3. INPUT LEVEL DISPLAY
4. RESET VOICE/FAX CHANNEL
5. **TERMINATE TEST**

FXO Telephone Interface

1. See the test hookup shown in Figure 5-3. Use the modular cable supplied. Connect one end to the modular jack on the voice/fax channel module labeled FXS/FXO. Connect the other end to an extension wall jack. In the example in Figure 5-3 this extension is labeled 123.

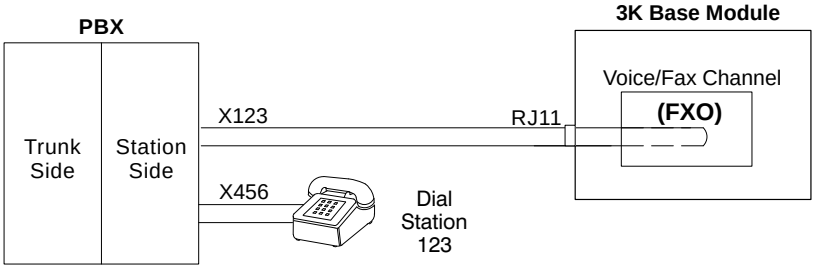


Figure 5-3. Example of a Switching FXO Loopback Test

2. Pick up any telephone extension (shown as 456 in Figure 5-3) and dial the number of the station being extended off-premises (shown as 123 in Figure 5-3).

3. Select the LOOPBACK option of the Voice Fax Tests Menu, and observe the indicators of the voice/fax channel being tested. The indicators should assume the following states:

OK	LO	RO	TM
0	1	0	1

Legend:
where 0 = OFF
1 = ON

4. Speak into the microphone. You should hear your own voice echoed back to you. It is normal to hear more than one echo of your voice.

Note: If you hear a busy tone, place the voice/fax channel temporarily into the FORCED OFF busyout mode (refer to 2-11). Be sure to restore this mode to SYSTEM CONTROLLED at the end of this test.

5. Hang up the telephone.

6. Stop the loopback test. To do that, select the TERMINATE TEST option. Verify that the LO and TM indicators on the channel being tested go off.

E&M Telephone Interface

1. See the test hookup shown in Figure 5-4. Connect one end of the RJ45 modular cable to the RJ1CX connector on 3K Base Module (see Figure 5-1). Connect the other end of the cable to the terminal block associated with the PBX.

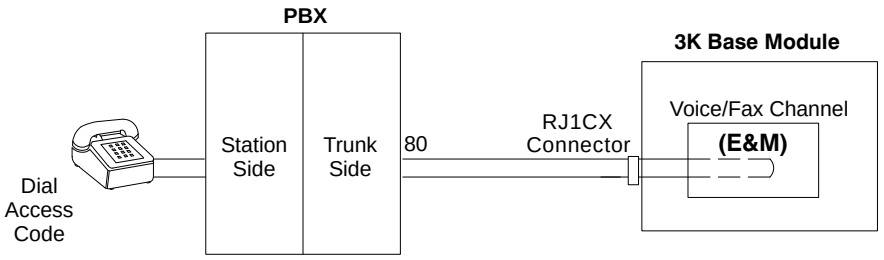
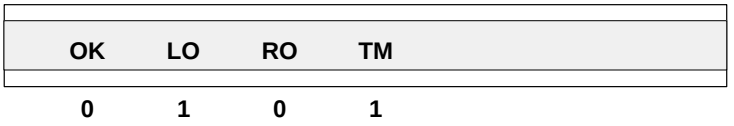


Figure 5-4. Example of a Switching E&M Loopback Test

2. Pick up any telephone extension, wait for the dial tone, then dial the trunk access code (typically 80).

3. Select the LOOPBACK option of the Voice Fax Tests Menu, and observe the indicators of the voice/fax channel being tested. The indicators should assume the following states:



Legend:
where 0 = OFF
1 = ON

4. Speak into the microphone. You should hear your own voice echoed back to you. It is normal to hear more than one echo of your voice.

Note: If you hear a busy tone, place the voice/fax channel temporarily into the FORCED OFF busyout mode (refer to 2-11). Be sure to restore this mode to SYSTEM CONTROLLED at the end of this test.

5. Hang up the telephone.

6. Stop the loopback test. To do that, select the TERMINATE TEST option. Verify that the LO and TM indicators on the channel being tested go off.

Loopback Tests, Force-Connect Mode

This series of tests is usually performed after contact with the remote end has been established, and requires the cooperation of an operator at each end. It can be used to test different segments of the connected voice/fax channel by alternating the end from which the test is initiated. In addition to echoing voice, it also tests the interaction between channels, such as the exchange of signalling and status information.

The following procedures assume that the voice/fax channels have been force connected (see page 2-11), and that the local voice/fax channel has been placed in loopback. All extension numbers and access codes are fictitious; the actual extension numbers are determined by your installation.

Note: When running a loopback test involving an AC15 interface, you may encounter intermittent background noise, especially when whistling into the microphone. This will occur only in the loopback mode, and does not affect the normal operation of your voice/fax channel.

VOICE/FAX TESTS
1. SELF TEST
2. **LOOPBACK**
3. INPUT LEVEL DISPLAY
4. RESET VOICE/FAX CHANNEL
5. TERMINATE TEST

FXS-to-FXS

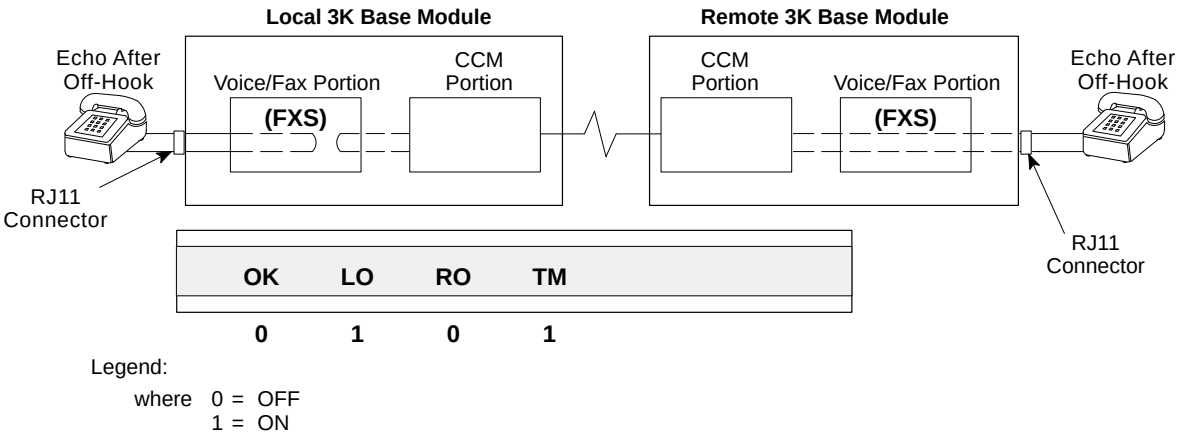


Figure 5-5. FXS-to-FXS Force Connect Test Setup

1. Establish a test setup as shown in Figure 5-5.
2. Lift the local receiver off-hook and speak into the microphone. You should hear your own voice echoed back. Place the local receiver on-hook.
3. At the remote end, lift the receiver off-hook and speak into the microphone. After a slight delay, you should hear your own voice echoed back.
4. Verify that the local indicators behave as in Figure 5-5. Place the remote receiver on-hook.
5. Stop the loopback test. To do that, select the TERMINATE TEST option. Verify that the LO and TM indicators on the local channel being tested go off.

Note: When the local telephone is lifted off-hook first, the remote telephone will ring. When the remote telephone is lifted off-hook first, the local telephone will not ring.

FXS-to-E&M

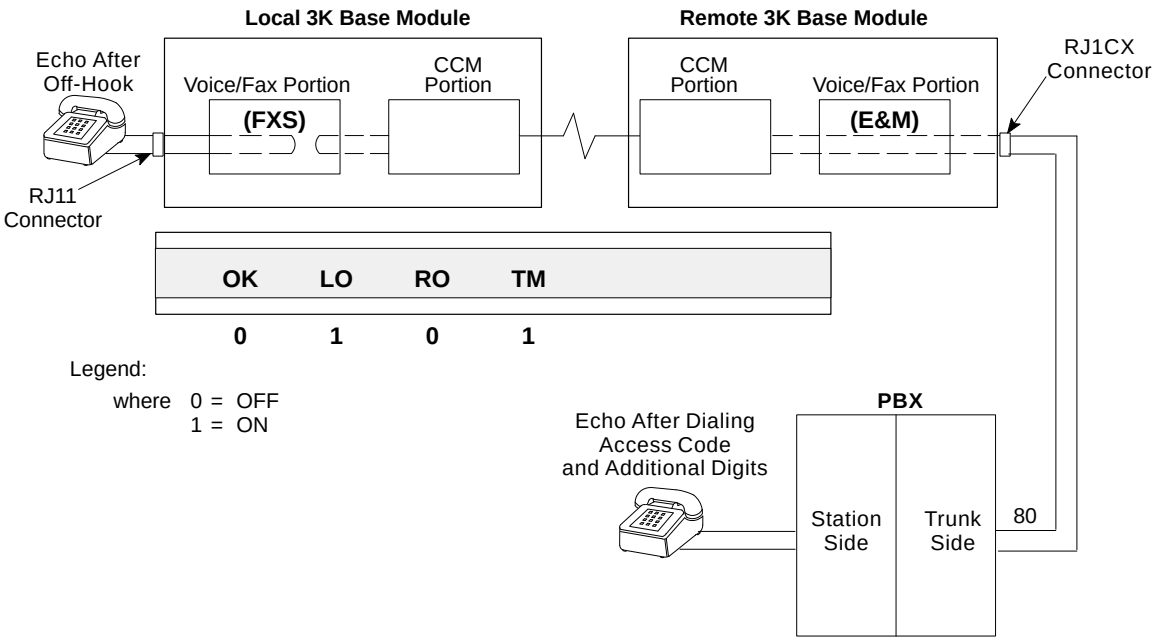


Figure 5-6. FXS-to-E&M Force Connect Test Setup

1. Establish a test setup as shown in Figure 5-6.
2. Lift the local receiver off-hook and speak into the microphone. You should hear your own voice echoed back. Place the local receiver on-hook.
3. At the remote end, lift the receiver and dial the access code. Wait approximately three seconds, then dial a random number of additional digits.
4. Speak into the microphone. After a slight delay, you should hear your own voice echoed back.
5. Verify that the local indicators behave as in Figure 5-6. Place the remote receiver on-hook.
6. Stop the loopback test. To do that, select the TERMINATE TEST option. Verify that the LO and TM indicators on the local channel being tested go off.

E&M-to-FXS

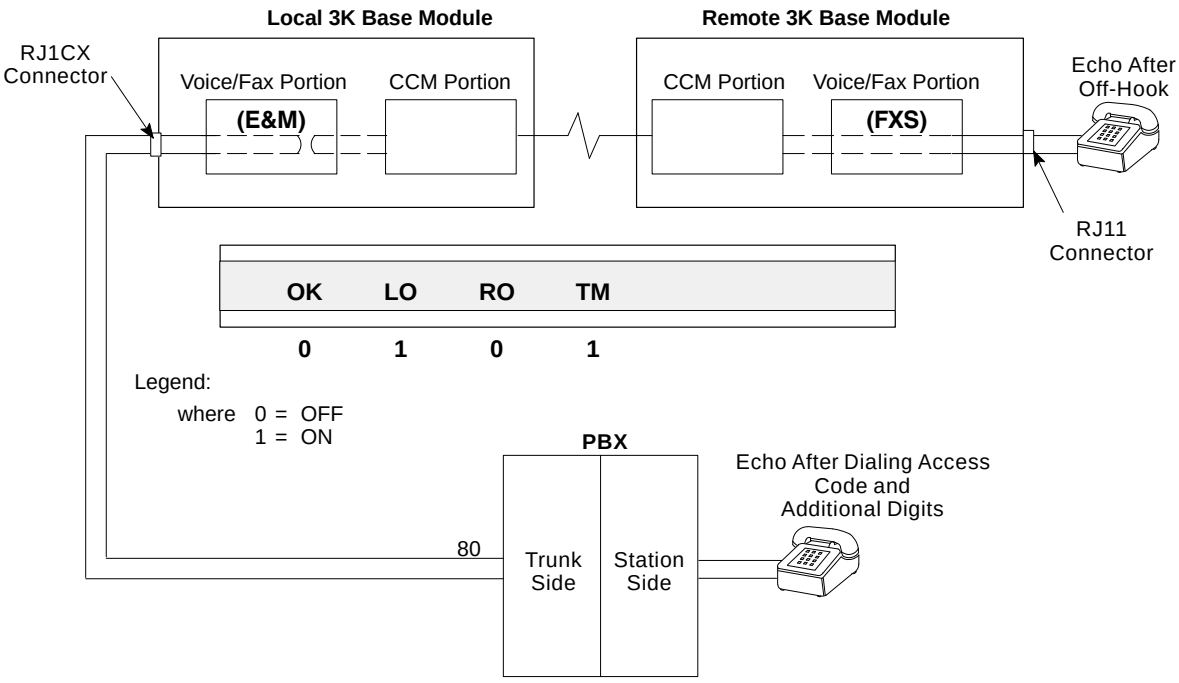


Figure 5-7. E&M-to-FXS Force Connect Test Setup

1. Establish a test setup as shown in Figure 5-7.
2. Lift the local receiver off-hook, and dial the access code. Wait approximately three seconds, then dial a random number of additional digits and speak into the microphone. You should hear your own voice echoed back. Place the local receiver on-hook.
3. At the remote end, lift the receiver off-hook. You should hear your own voice echoed back.
4. Verify that the local indicators behave as in Figure 5-7. Place the remote receiver on-hook.
5. Stop the loopback test. To do that, select the TERMINATE TEST option. Verify that the LO and TM indicators on the local channel being tested go off.

E&M-to-E&M

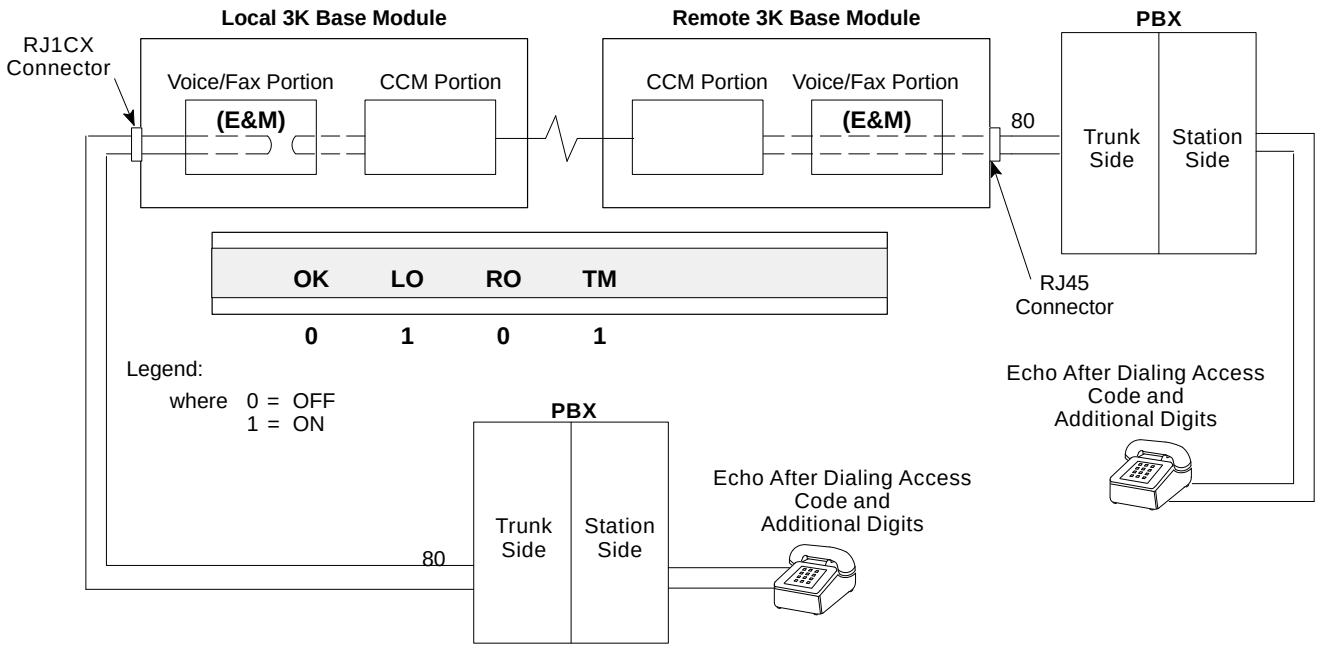


Figure 5-8. E&M-to-E&M Force Connect Test Setup

1. Establish a test setup as shown in Figure 5-8.
2. Lift the local receiver off-hook, and dial the access code. Wait approximately three seconds, dial a random number of additional digits and speak into the microphone. You should hear your own voice echoed back. Place the local receiver on-hook.
3. At the remote end, lift the receiver and dial the access code. Wait approximately three seconds and dial a random number of additional digits.
4. Speak into the microphone. After a slight delay, you should hear your own voice echoed back.
5. Verify that the local indicators behave as in Figure 5-8. Place the remote receiver on-hook.
6. Stop the loopback test. To do that, select the TERMINATE TEST option. Verify that the LO and TM indicators on the local channel being tested go off.

FXS-to-FXO

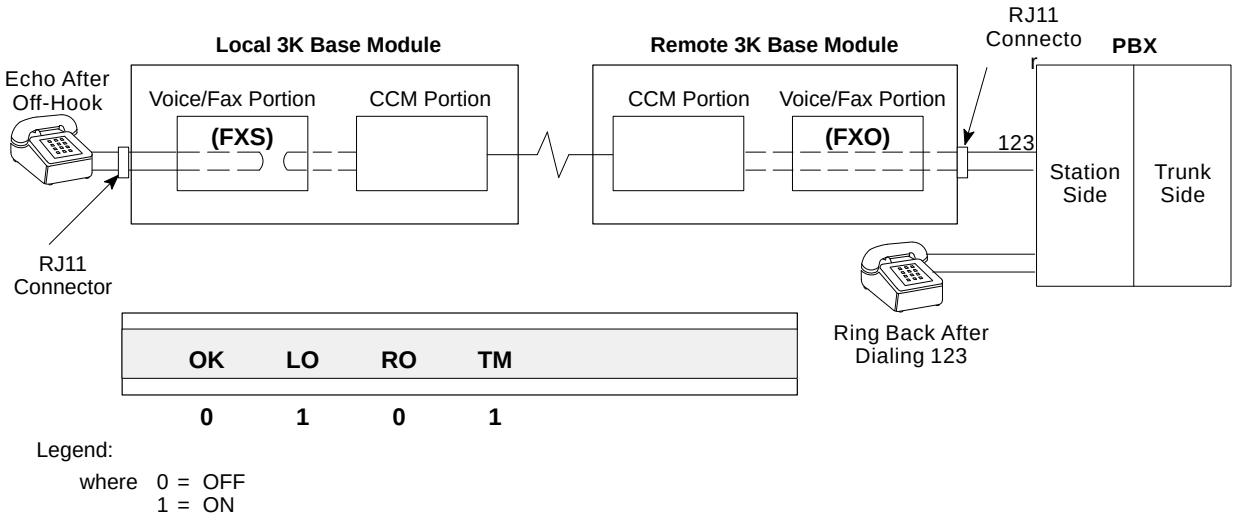


Figure 5-9. FXS-to-FXO Force Connect Test Setup

1. Establish a test setup as shown in Figure 5-9.
2. Lift the local receiver off-hook and speak into the microphone. You should hear your own voice echoed back. Place the local receiver on-hook.
3. At the remote end, lift the receiver and dial extension 123. You will hear a ringback at the remote end; the local end will not ring.
4. Lift the local receiver off-hook.
5. At the remote end, speak into the microphone. After a slight delay, you should hear your own voice echoed back.
6. Verify that the local indicators behave as in Figure 5-9. Place the remote receiver on-hook.
7. Stop the loopback test. To do that, select the TERMINATE TEST option. Verify that the LO and TM indicators on the local channel being tested go off.

FXO-to-FXS

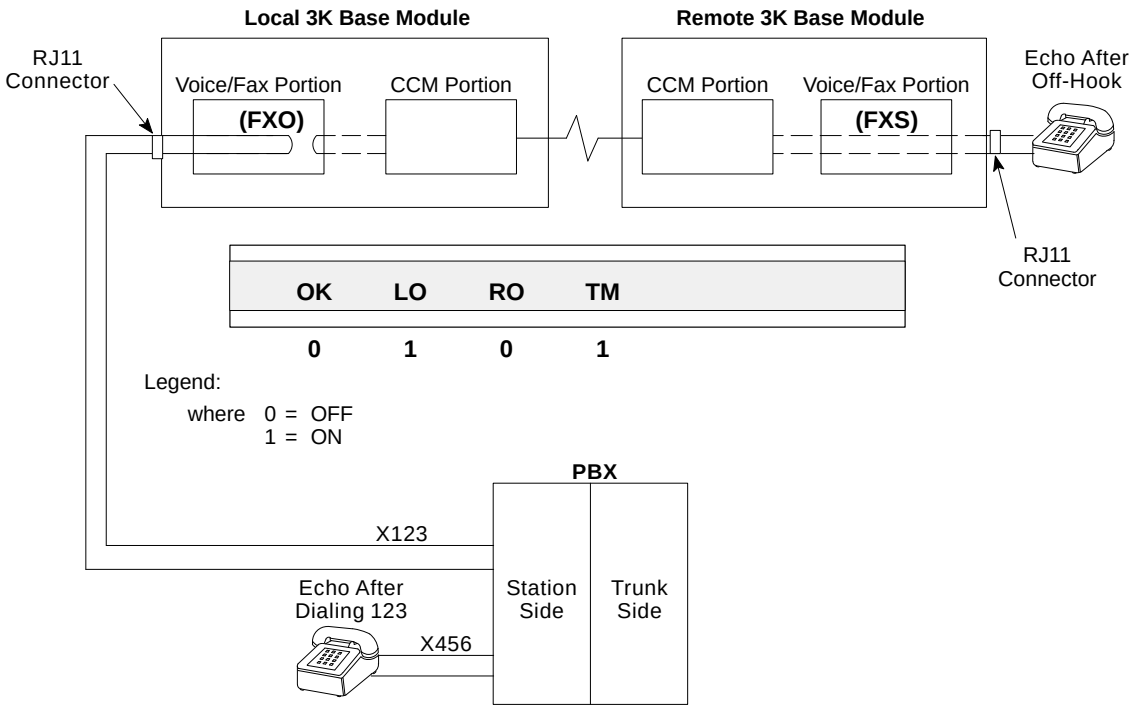


Figure 5-10. FXO-to-FXS Force Connect Test Setup

1. Establish a test setup as shown in Figure 5-10.
2. At the local end, lift the receiver off-hook and dial extension 123. You may hear a momentary ringback.
3. Wait approximately 4 seconds, then speak into the microphone. You should hear your own voice echoed back. Place the local receiver on-hook.
4. At the remote end, lift the telephone off-hook and speak into the microphone. After a slight delay, you should hear your own voice echoed back.
5. Verify that the local indicators behave as in Figure 5-10. Place the remote receiver on-hook.
6. Stop the loopback test. To do that, select the TERMINATE TEST option. Verify that the LO and TM indicators on the local channel being tested go off.

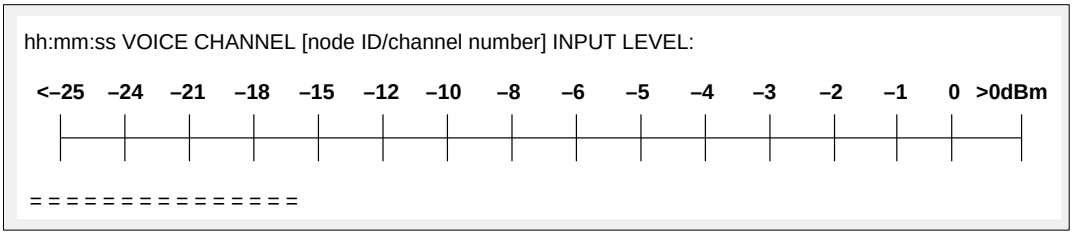
Input Level Display

```
VOICE/FAX TESTS
        1.  SELF TEST
        2.  LOOPBACK
        3.  INPUT LEVEL DISPLAY
        4.  RESET VOICE/FAX CHANNEL
        5.  TERMINATE TEST

CR  –  ACCEPT ENTRY
M   –  MAIN MENU

ENTRY;
```

The INPUT LEVEL DISPLAY option of the Voice/Fax Tests Menu allows you to monitor the level of the incoming voice signal, a useful tool when diagnosing voice quality problems.

This test is non disruptive and can be performed without affecting voice operation. It is also useful when installing a channel to verify a proper connection to the associated telephone equipment.

When the Command Facility terminal is operating at a data rate of 9600 bps or higher, the input level displayed will be as shown in Figure 5-11.



where = is a dynamic value controlled by the measured voice level

Figure 5-11. Dynamic Input Level Display

To stop the dynamic display, press the carriage return <cr>. This will get you back to the Voice/Fax Test Menu.

If the asynchronous terminal data rate is lower than 9600 bps, or if the Command Facility Main Menu was accessed via $CMD rather that through the Command Port, a typical display will be as follows:

```
VOICE/FAX CHANNEL [SIMI/E1] INPUT LEVEL -10 dB
```

The voice/fax channel indicators LO, RO, and TM also form a bar graph display as shown in Figure 5-12. To terminate the input level display test, select option 5, Terminate Test.

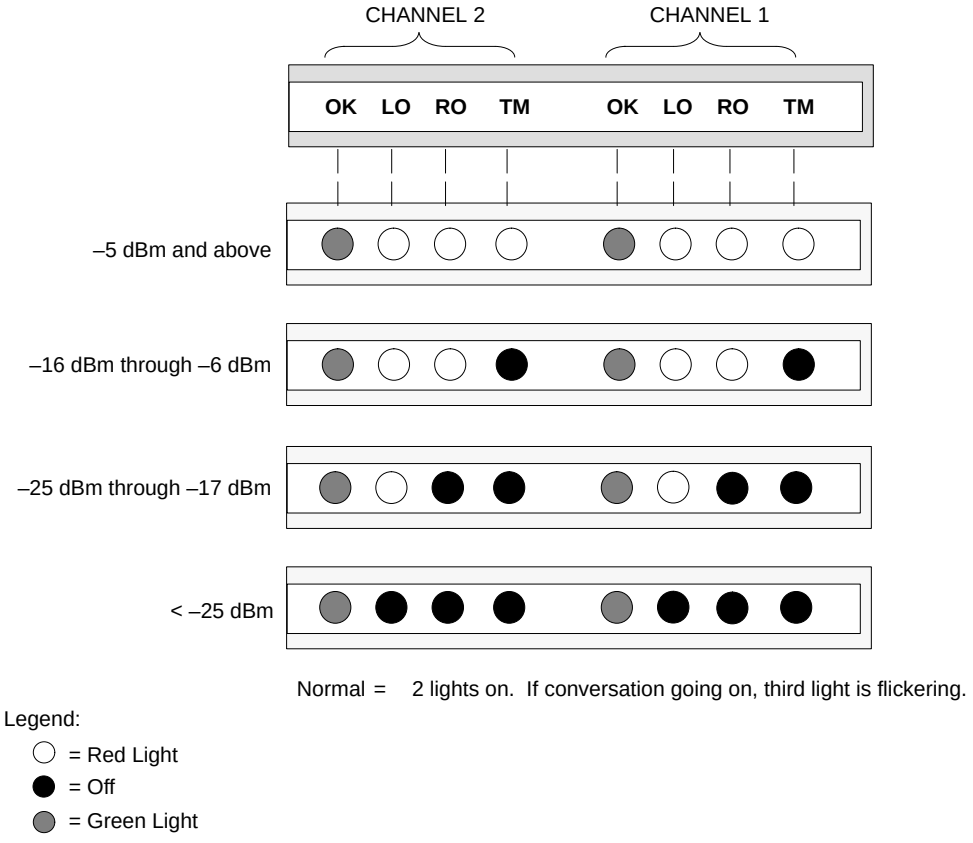


Figure 5-12. Voice/Fax Channel Bar Graphs

Reset Voice/Fax Channel

```
VOICE/FAX TESTS
        1. SELF TEST
        2. LOOPBACK
        3. INPUT LEVEL DISPLAY
        4. RESET VOICE/FAX CHANNEL
        5. TERMINATE TEST
```

The RESET VOICE/FAX CHANNEL option of the Voice/Fax Tests Menu is a means for resetting either voice/fax channel individually without affecting the operation of the other channel.

Testing of Two Local Voice/Fax Channels

Any two channels may be tested locally without using the WAN link. To do that, you simply connect them as you normally would; either by dialing an extension number (switched), or using the force connect method described on page 2-29. Then, perform the same voice tests as those described on pages 5-4 through 5-14.

Figure 5-13 is an example of a switched hookup between two local voice/fax channels in an FXO-to-FXS application. For clarity, PBX extensions are assigned 3-digit numbers and voice/fax channels 4-digit numbers. All extension numbers are fictitious; the actual extension numbers are determined by your installation.

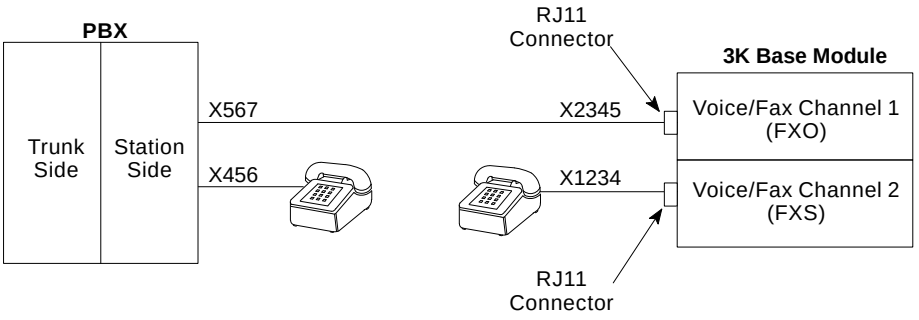


Figure 5-13. Connecting Two Local Voice Fax Channels

Specifications 6

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E&M Telephone Interface Analog Specifications 6-3
FXS Telephone Interface Analog Specifications 6-5
FXO Telephone Interface Analog Specifications 6-7

This section contains specifications associated with the voice/fax portion of the 3K Base Module.

General

Number of Voice/Fax Channels:	2
Signals Supported:	Analog voice, Group 3 facsimile (fax), and modem
Fax Signal Types:	V.21 channel 2, V.27 ter 2400, V.27 ter 4800, V.29 7200, and V.29 9600
Modem Signal Types:	V.22 and V.22 bis
Modem Signal Sensitivity:	–43 dBm

Telephone Interfaces

PBX Tie Trunk:	E&M Types I, II and V, 2-wire or 4-wire
PBX Station or Central Office/PSTN:	FXO Loop Start, 2-wire
Key Telephone Systems, Telephone Set, or PBX CO Trunk:	FXS Loop Start, 2-wire
Interface Connectors Per Channel:	RJ1CX modular jack for FXS and FXO type interfaces, and RJ1CX modular jack for E&M type interfaces
Indicators per Channel:	OK, Local Off-Hook, Remote Off-Hook, and Test
Diagnostics:	Self-Test, Loopback, Input Level Display

E&M Telephone Interface Analog Specifications

Input Impedance	
4-wire (default):	600 Ω ± 10% resistive only
2-wire:	Resistive or complex, as strapped
Resistive:	600 Ω ± 10%
Complex:	370 Ω + 0.31 μf //[1] 620 Ω or 220 Ω + 0.12 μf // 820 Ω
Input Level Gain:	Adjustable from –6 dB to +19 dB in 1 dB increments
Output Level Attenuation:	Adjustable from 0 dB to 25 dB in 1 dB increments
Maximum Output Level, 4-wire only:	+7 dB
Input and Output Level Adjustment Restrictions:	If your equipment is operated in the U.S.A. or Canada, input and output levels must be set to 0 dBm.
Longitudinal Balance	
1 to 1004 Hz:	≥ 58 dB
1005 Hz to 3404 Hz:	≥ 48 dB
Insertion Loss at 1004 Hz	
2-wire T/R[2] of Port 1 to 2-wire T/R of Port 2:	2 dB nominal
4-wire T/R of Port 1 to 4-wire T1/R1 of Port 2:	2 dB nominal
Return Losses	
Echo:	≥ 22 dB
Singing:	≥ 18 dB
204 to 3400 Hz:	≥ 16 dB
Frequency Response Over the Range of 304 Hz to 3404 Hz:	+1 dB/–2 dB with respect to 1004 Hz

[1] // indicates in parallel with.
[2] Transmit/Receive.

Idle Channel Noise	
Inband	
"C" Message:	≤ 20 dBrnc
Psofometric:	≤ –65 dBmpo
Out of band, 10 KHz to 10 MHz Transverse or Metallic Noise:	≤ –70 dBm
Longitudinal Noise:	≤ –65 dBm
Non Linear Distortion, (Multitone Signal) Second and Third Harmonics:	≤ 40 dB below signal level
Signal to Noise Ratio at 1004 Hz:	≥ 37 dB
Echo Suppression: (2-wire):	≥ 35 dB
Echo Canceling: (2-wire):	≤ 16 ms or ≤ 1000 Km (600 miles)
Signaling	
E-Lead Current Limit:	≤ 25 ma
M-Lead Sensitivity:	48V in series with ≤ 1400 Ω
DC Pulse Distortion:	≤ 3% at 10 pps
Signaling Formats	
AC:	DTMF & 2280 Hz (AC15, 4-wire)
DC:	Steady or Pulsed
Signaling Types:	I, II, and V, as strapped

FXS Telephone Interface Analog Specifications

Input Impedance:	600 Ω or Complex, as strapped
600 Ω (Default):	600 Ω in series with a 2.15 μf capacitor
Complex:	370 Ω + 0.31 μf //[1] 620 Ω or 220 Ω + 0.12 μf // 820 Ω
Insertion Loss: (End to End)	2 dB nominal @ 1004 Hz between Transmit and Receive of Port 1 and Transmit and Receive of Port 2.
Frequency Response Over the Range of 304 Hz to 3404 Hz:	+1 dB/–2dB with respect to 1004 Hz
Return Losses	
Echo Return Loss:	≥ 22 dB
Singing Return Loss:	≥ 18 dB
204 to 3404 Hz	≥ 16 dB
Input Level Gain:	Adjustable from –6 dB to +19 dB in 1 dB increments
Output Level Attenuation:	Adjustable from 0 dB to 25 dB in 1 dB increments
Input and Output Level Adjustment Restrictions:	If your equipment is operated in the U.S.A. or Canada, input and output levels must be set to 0 dBm.
Longitudinal Balance	
200 to 1004 Hz:	≥ 58 dB
1004 to 3404 Hz:	≥ 48 dB
Non-Linear Distortion (Multitone Signal) Second and Third Harmonics:	≤ 40 dB below signal
Signal to Noise Ratio at 1004 Hz:	≥ 37 dB
Echo Suppression:	≥ 35 dB
Echo Canceling:	≤ 16 ms (≤ 1000 Km or 600 miles)
Crosstalk (Near/Far End) Between Channels:	≤ 75 dB

[1] // indicates in parallel with.

Signaling Formats	
AC:	DTMF
DC:	Pulsed
DTMF Distortion:	≤ 1.5% (transparently passed)
Battery current into 130 Ω load:	≥ 36 milliamperes
Battery current into 900 Ω total loop:	20 milliamperes minimum
DC Loop Range:	1750 Ω or 9600 feet (2926 meters) on 24 AWG wire
Ringing Tone (Software Selectable):	25 Hz (default), or 50 Hz
Ringing Voltage, with Ac Load of 2 Ringers (4000 W) and zero line	
25 Hz Tone:	≥ 54 VRMS
20 and 50 Hz Tones:	≥ 50 VRMS
Ringing Current:	≥ 15 milliamperes into 5000 Ω load
Ring Cadence	
Repeated ring:	Matches ringing cadence from remote voice/fax channel
Interrupted ring:	On for 2 seconds, off for 4 seconds.

FXO Telephone Interface Analog Specifications

Input Impedance:	600 Ω or Complex, as strapped.
600 Ω (Default):	600 Ω in series with a 2.15 μf capacitor
Complex:	370 Ω + 0.31 μf //[1] 620 Ω or 220 Ω + 0.12 μf // 820 Ω
Insertion Loss: (End to End)	2 dB nominal @ 1004 Hz between Transmit and Receive of Port 1 and Transmit and Receive of Port 2.
Input Level Gain:	Adjustable from –6 dB to +19 dB in 1 dB increments
Output Level Attenuation:	Adjustable from 0 dB to 25 dB in 1 dB increments
Input and Output Level Adjustment Restrictions:	If your equipment is operated in the U.S.A. or Canada, input and output levels must be set to 0 dBm.
Return Losses	
Echo:	≥ 22 dB
Singing:	≥ 18 dB
200 to 3400 Hz:	≥ 16 dB
Longitudinal Balance	
204 to 1004 Hz:	≥ 58 dB
1004 to 3404 Hz:	≥ 48 dB
Frequency Response Over the Range of 304 Hz to 3404 Hz	+1 dB/–2 dB with respect to 1004 Hz
Idle Channel Noise	
Inband	
"C" Message:	≤ 20 dBrnc
Psofometric:	≤ –65 dBmpo
Out of band:	10 KHz to 10 MHz
Transverse or Metallic Noise:	≤ –70 dBm
Longitudinal Noise:	≤ –65 dBm
Non Linear Distortion, (Multitone Signal) Second and Third Harmonics:	≤ 40 dB below signal level
Signal to Noise Ratio at 1004 Hz:	≥ 37 dB
Echo Suppression:	≥ 35 dB
Echo Canceling:	≤ 16 ms or ≤ 1000 Km (600 miles)

[1] // indicates in parallel with.

Crosstalk (Near/Far End) Between Channels:	≤ 75 dB
Signaling Formats	
AC:	DTMF
DC:	Pulsed
DTMF Distortion:	≤ 1.15% (transparently passed)
Pulse Distortion:	≤ 3 %
DC Loop Range	
48 V Battery:	≤ 1750 Ω
42.5 V Battery:	≤ 1500 Ω
Disconnect Supervision (Software Selectable)	
Tone:	Voice/fax transmission will be disconnected in response to a call progress tone of less than 600 Hz.
Power Interrupt:	Voice/fax transmission will be disconnected in response to a pulse of 600 ms minimum.
Off-Hook DC V/I Characteristics at Tip-Ring:	≤ 300 Ω
Ringing Voltage Input, 18 to 53 Hz	
Detect:	≥ 25 VRMS
Not Detect:	≤ 15 VRMS
Ringing Cadence Repeat Distortion:	≤ 2 %

Agency Requirements A

Information provided in the following paragraphs is applicable only to units installed in the U.S.A. and Canada. If your installation is located outside of the U.S.A. and Canada, you must meet the requirements of the country wherein the equipment is operated.

FCC Requirements

The Federal Communications Commission (FCC) has approved this equipment for operation in the U.S.A. In so doing, the FCC approves this equipment as not being harmful to the telephone network when this equipment is connected directly to the telephone lines.

Requirements for Operation with the Telephone Company

User's Responsibility

If a need arises in the future, the telephone company will call the user and request the following information:

- Manufacturer of the device: BLACK BOX Corp.
- Model number of the device (s): Marathon 3K Model 5003B
- FCC Part 68: BEBUSA 75539-PX-N
- Type of Interfaces: E&M, FXO, or FXS, as strapped
- Jacks, Facility Interface Codes, Service Order Codes, and Ringer Equivalence. See below.

Interface Type	USO Jack	FIC	SOC	RE
E&M	RJ1CX	TL11M/E 2-Wire	–	–
		TL12M/E 2-Wire	–	–
		TL31M/E 4-Wire	–	–
		TL32M/E 4-Wire	–	–
OPS (FXS)	RJ11C/W	OL13C 2-Wire	9.0F	
OPX (FXO)	RJ11C/W	O2LS2 2-Wire	9.0F	0.98

FIC = Facility Interface Code
SOC = Service Order Code
RE = Ringer Equivalence

Telephone Company Rights and Responsibilities

If your equipment causes harm to the Telephone Network, the telephone company may discontinue your service temporarily. If possible, they will notify you in advance. But if advance notice is not practical, you will be notified as soon as possible. You will be given the opportunity to correct the situation and you will be informed of your right to file a complaint with the FCC.

Your telephone company may make changes in its facilities, equipment, operation, or procedures that could affect the proper functioning of your equipment. If they do, you will be notified in advance to give you an opportunity to maintain uninterrupted service.

Repair Instructions

If you experience any operational problems while using your equipment, determine if the problem is due to a malfunction in your equipment or in the telephone interface.

> ⚠ ***WARNING***
>
> Do not attempt to repair the Marathon 3K Base Module. Attempts to repair the Marathon 3K Base Module may cause injury, and may also damage equipment on the telephone network. Attempts to repair the Marathon 3K Base Module are violations of FCC rules. Repair to the Marathon 3K Base Module can only be made by the manufacturer, its authorized agents, and by others who may be authorized by the FCC. Please contact BLACK BOX Corp. Customer Service hotline at (724) 746-5500 for further information.

1. **The problem may be in the telephone service**

   Perform the appropriate loopback tests given in Section 5 to verify the integrity of the voice/fax portion of the Marathon 3K Base Module. If the tests fail, disconnect your equipment from the telephone interface and notify the telephone company of the problem.

2. The problem may be in your equipment.

If the loopback tests indicate that the telephone line is operational, or if the telephone line is known to work with other equipment, then the problem is most likely in your Marathon 3K Base Module. To check your Marathon 3K Base Module, follow the test procedures given in Section 5.

Refer to the following guidelines for obtaining service:

a. Verify that the Marathon 3K Base Module is configured (see Section 2) and strapped (see the 3-Slot Chassis Installation and Cabling Manual, part number 800-1846) as required, and that you are using the proper cables.

b. If the Marathon 3K Base Module is covered by BLACK BOX's Warranty, follow the procedure set forth in the Warranty (see the Warranty page) for obtaining repair or replacement of the Marathon 3K Base Module.

c. If the Marathon 3K Base Module is no longer covered by the Warranty, contact Black Box Customer Sales representative

BABT Notes

Ringer Equivalence Number (REN, U.K.)

The Ringer Equivalence Number (REN) for the FXO PSTN port is 1.0.

Satisfactory operation of this equipment will be achieved if the sum of RENs of terminal equipment connected in parallel does not exceed 4. For the purposes of this calculation the REN of a terminal equipment rented or bought from plc should be assumed to be REN = 1, unless otherwise specified.

Loop Disconnect Dialing

Although this equipment can use either loop disconnect or DTMF dialing, only the performance of the DTMF signaling is subject to regulatory requirements for the correct operation. It is therefore strongly recommended that the equipment be set to use DTMF signaling for access to public or private emergency services. DTMF signalling also provides faster call setup.

Equipment Attachment Limitations for Operation in Canada

CP–01, **Part I**

Section 10.1

NOTICE: The Canadian Department of Communications label identifies certified equipment. This certification means that the equipment meets certain telecommunications network protective, operational and safety requirements. The Department does not guarantee the equipment will operate to the user's satisfaction.

Before installing this equipment, users should ensure that it is permissible to be connected to the facilities of the local telecommunications company. The equipment must also be installed using an acceptable method of connection. In some cases, the company's inside wiring associated with a single line individual service may be extended by means of a certified connector assembly (telephone extension cord). The customer should be aware that compliance with the above conditions may not prevent degradation of service in some situations.

Repairs to certified equipment should be made by an authorized Canadian maintenance facility designated by the supplier. Any repairs or alterations made by the user to this equipment, or equipment malfunctions, may give the telecommunications company cause to request the user to disconnect the equipment.

Users should ensure for their own protection that the electrical ground connections of the power utility, telephone lines and internal metallic water pipe system, if present, are connected together. This precaution may be particularly important in rural areas.

CAUTION

Users should not attempt to make such connections themselves, but should contact the appropriate electric inspection authority, or electrician, as appropriate.

CP–01, **Part I,**

Section 10.2

NOTICE: The **Load Number** (LN) assigned to each terminal device denotes the percentage of the total load to be connected to a telephone loop which is used by the device, to prevent overloading. The termination on a loop may consist of any combination of devices subject only to the requirement that the total of the Load Numbers of all the devices does not exceed 100.

Voice/Fax Channel Level Adjustment Procedure B

This Appendix contains procedures for adjusting the input/output levels of the voice/fax channel.

These procedures are intended for equipment operated outside of the U.S.A. and Canada, and must meet the standards of the country wherein the equipment is used.

Overview

The input/output levels of the voice/fax channel are set at the factory for a default value of 0/0 dB. At some installations, the input/output levels need to be adjusted to compensate for external network losses and variations, to optimize the voice compression operation, and to provide a consistent communication level throughout the network. However, no adjustment within the voice/fax circuits can correct for voice distortions prior to the arrival at the input of the voice port, or for unusual impedances inherent in the external network.

The objective of these procedures is to obtain an input level of –4 dBm, and achieve an overall circuit loss ranging from 8 to 16 dB without introducing an input gain setting of more than 6 dB to achieve it. An excessive input gain setting may cause echoes or other voice impairments such as singing or repeated DTMF signaling digits.

Adjust all Voice Ports (Channels) at One Location

If yours is a dual channel voice/fax facility, both voice channels may be adjusted by a single technician and without going across the network. This is accomplished by force connecting the two voice channels, or by placing a switched call between them.

First, a signal from a fixed dBm source is applied in one direction through the connected pair, with the voice channel nearest the signal source serving as the sending channel, and the other voice channel serving as the receiving channel. Adjustment involves setting the sending channel input level to a Transmission Level Point (TLP) of a –4 dBm input level display, and the receiving channel's output level to a reading of –8 dBm or lower at the associated dB meter (see Figure B-1).

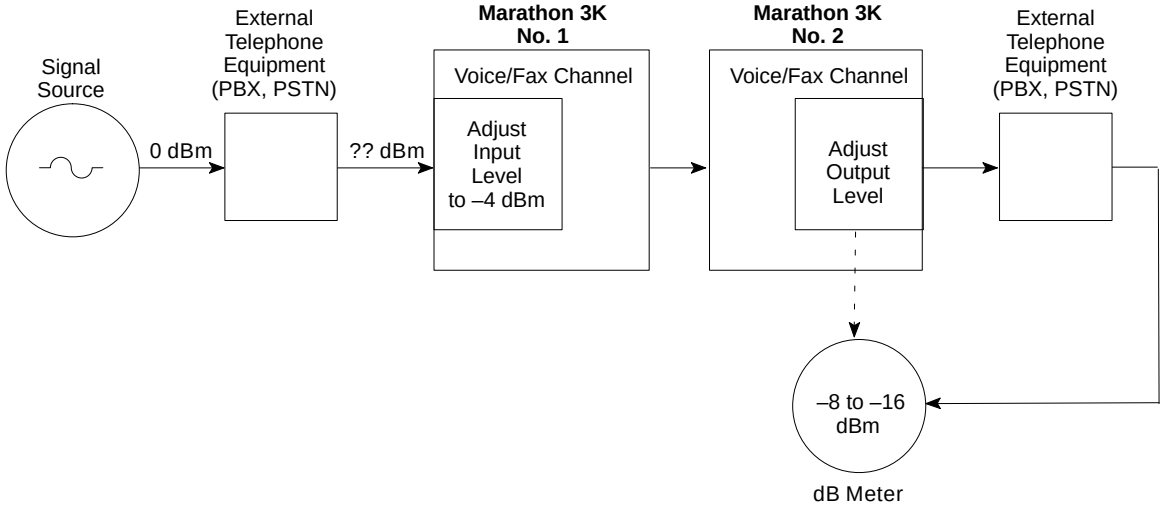


Figure B-1. Basic Adjustment Objective

Next, the two voice channels remain connected as before, but the signal flow is reversed. What was previously the receiving voice channel now becomes the sending channel, and vice versa, and the adjustments are made in a reverse order.

If yours is a single voice/fax channel facility, you must pair up this channel with another, compatible voice channel across the network. In this case the adjustment procedure requires two technicians, one at each location.

If your Marathon 3K includes, in addition to the built-in voice channels, another voice/fax module configured for the same telephone interface, you may copy the results into that channel.

Private Network or PSTN

The voice channels were designed for communication among multiple offices of a single company. As such, the levels obtained from the various telephone instruments of that private Integration network will normally be consistent within a few dBs. Under these conditions, the input level adjustment procedure is based on a signal source (speaker) of a 1004 Hz tone at 0 dBm.

Where regulations allow, the voice channels may connect directly or indirectly to the Public Switched Telephone Network (PSTN), presenting a different set of conditions. Here, the calls may originate from widely different sources with extreme variations in signal levels (up to 20 dB). Further, the levels from calls within an office will often be different from the PSTN levels.

Therefore, it is strongly recommended that voice channels be dedicated *either* for private network use *or* PSTN use, but not both.

If the planned usage for a single voice channel is both private network and PSTN operation, then one of the applications should be selected as the primary use and the voice channel aligned for that purpose. The secondary application should be validated to determine if a compromise setting is required.

Equipment Required

1. Two MetroTel Voice Network Analyzers (VNA-70A) or equivalent, capable of generating a 1004 Hz tone at 0 dBm and also capable of measuring a receive tone level in dBm units.
2. One asynchronous ASCII Terminal, e.g., one normally attached to the Marathon 3K unit. An alternative is a Personal Computer running a Terminal Emulation software package such as ProComm+.
3. One Telephone Lineman's Testset (Butt Set). If not available, a standard single-line analog telephone instrument may be substituted. A second phone or testset may be helpful.

Preliminary Considerations and Connections

The telephone equipment to which a voice channel is attached should be installed and made functional. If the telephone equipment is not installed, then the installation should be scheduled after the telephone equipment is installed and configured or a joint installation should be scheduled (as a last resort).

Make sure that an asynchronous ASCII terminal or equivalent is connected to an asynchronous port on the Marathon 3K unit. This terminal will be used for configuration and testing in the subsequent adjustment procedures.

Interface Pairings

As described previously, every adjustment procedure described here involves a local pair of force-connected voice channels. The voice pairings treated here follow the most common voice applications, as follows:

- PBX Trunk Application (E&M to E&M or FXS to FXS)
- PBX or PSTN Station Application (FXO to FXS)
- Key Telephone System Application (FXS to FXS)

Preliminary Settings

Before attempting to adjust the input output levels, configure them to the setting shown in Table B-1.

Table B-1. Preliminary Settings

Application	Input Level Setting	Output Level Setting
PBX Trunk (E&M)	–2	2
PBX Station (FXO)	2	0
Key Telephone Systems (FXS)	–3	2

Checking the Operation of the PBX Station

This procedure is used to verify that the losses across the PBX, in a PBX station application, are within acceptable limits for operation with the voice network.

1. Use the test setup shown in Figure B-2. Initially, set both Voice Network Analyzers (VNA) to BRDG, TEST and OFF. The extension numbers in Figure B-2 are fictitious, used here for reference only.
2. Lift the calling telephone set off-hook, wait for the dial tone and dial station 5678. The receiving telephone will ring.
3. Place the SETUP switch on the receiving Voice Network Analyzer to TERM. The telephone will stop ringing.
4. Place the SETUP switch on the calling Voice Network Analyzer to TONE, and disconnect the calling telephone from the Voice Network Analyzer.
5. Observe the displayed value on the receiving Voice Network Analyzer. It should read between –5 dBm and –8 dBm. If the level is outside of this range, the PBX should be checked for proper operation by authorized service personnel.

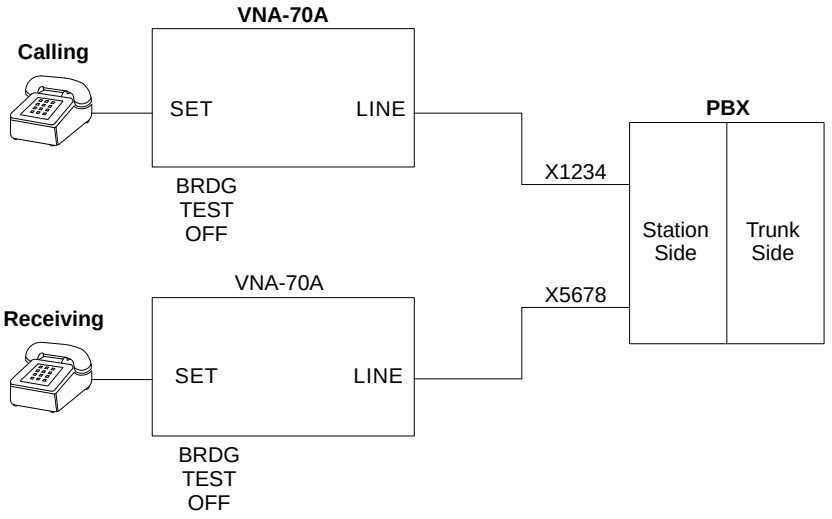


Figure B-2. Checking the PBX in a Station Application

PBX Tie Trunk Application Adjustments

Many PBXs have selectable input/output level combinations for Tie Trunks, e.g., +7/–16, –16/+7 and 0/0 dB. The 0/0 dB level option should be selected. Also, PAD (2 dB) Switching Option, if available, should be selected.

1. Use the test setup shown in Figure B-3. Initially, set both Voice Network Analyzers (VNA) to BRDG, TEST and OFF. The extension numbers in Figure B-3 are fictitious, used here for reference only.

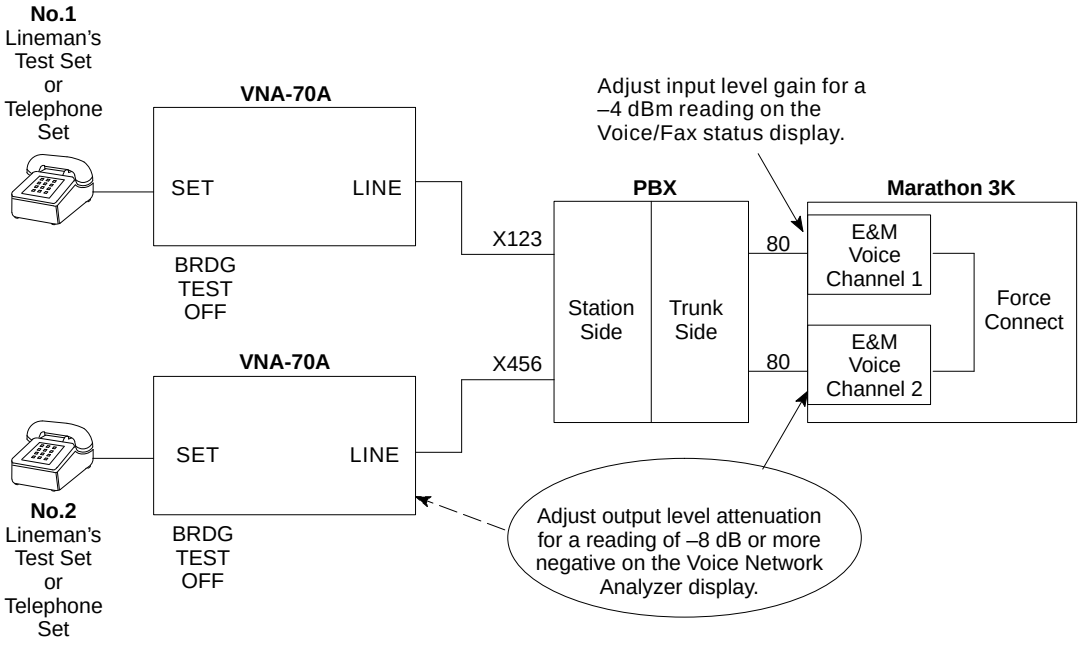


Figure B-3. PBX Tie Trunk Application Test Setup

2. Connect two E&M voice channels within the Marathon 3K. (Use either force connect or switched call). Set any additional voice channels to FORCE ON busy condition to insure that the test calls operate only through the channels under test. It is assumed that the signalling format for the E&M voice channel is configured for DC.
3. Place a call from the No.1 telephone to the No.2 telephone. To do that, lift the No.1 telephone off-hook, wait for the dial tone and dial the tie trunk access code (typically 80). A second dial tone should be heard from the PBX through the force connected channels.
4. Dial extension 456. The No.2 telephone set will ring.
5. Place the SETUP switch on the No.2 Voice Network Analyzer to TERM. The telephone will stop ringing.
6. Place the SETUP switch on the No.1 Voice Network Analyzer to TONE, and disconnect the No.1 telephone from the Voice Network Analyzer.

7. Using the asynchronous terminal, enter the Voice/Fax Channel Status. To do that, follow this path:

Command Facility Main Menu

Status/Statistics

↓

Voice/Fax Status

↓

ENTER NODE-ID/CHANNEL # (OR RANGE OF CHANNELS) (^X TO ABORT):

8. Observe the display on the asynchronous terminal, determine which channel is receiving a tone, and note its input level. If the reading is more positive than –4 dBm, apply negative gain (input attenuation) in the amount that this reading is above –4 dBm.

Example: The reading is –1 dBm.

The Input Level Gain Setting in the Voice/Fax Channel Characteristics Menu is at position 5 (–2 dB).

Add three increments of negative gain by moving to position 2, –5 dB.

To adjust the input level, enter the Input Level Gain Menu for the appropriate voice/fax channel number, by following these steps:

Command Facility Main Menu

Configure Local Nodes

↓

Voice/Fax Menu

↓

Voice/Fax Channel Configuration

↓

ENTER NODE-ID/CHANNEL # (^X TO ABORT:)

↓

Voice/Fax Channel Characteristics

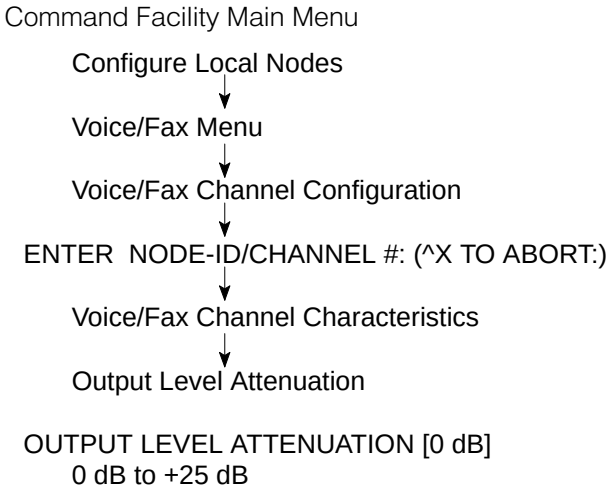
↓

Input Level Gain

INPUT LEVEL GAIN [0 dB]

After the input level gain adjustment is completed, jot down the setting used to bring the input level to –4 dBm. You will need it for comparison when setting other E&M channels.

9. Move to the voice/fax Output Level Attenuation configuration menu for the E&M channel No. 2 (the channel that is connected to the channel just adjusted) by following these steps:



10. Read the value of the receive level on the No.2 Voice Network Analyzer display. The receive level should fall between –8 and –10 dBm. Set the output attenuation until the receive level display is –8 dBm (e.g., the output attenuation will be between 0 and 2 db). If the receive level is below (more negative than) –10 dBm, reduce the output attenuation until no further adjustment is possible (0 attenuation). Perceived voice quality is not significantly affected until the receive level drops below –16 dBm.

    You have now established the input level gain setting on the No.1 channel, and the output level attenuation setting on the No.2 channel. Next, adjust the input level gain setting on the No.2 channel, and the output level attenuation setting on the No.1 channel.

11. Reconnect the No.1 telephone set to its Voice Network Analyzer, and take it off-hook.
12. Set the SETUP switch on No.1 Voice Network Analyzer to TERM, then disconnect No.1 telephone.
13. Take the No.2 telephone off-hook.
14. Place the SETUP switch on the No.2 Voice Network Analyzer to TONE.
15. Disconnect the No.2 telephone from the Voice Network Analyzer.
16. Using the asynchronous terminal, enter the Voice/Fax Channel Status as in step 7.

17. Observe the display on the asynchronous terminal, determine which channel is receiving a tone, and note its input level. If the reading is more positive than –4 dBm, apply attenuation (negative gain) in the amount that this reading is above –4 dBm.

    To adjust the input level, enter the Input Level Gain Menu for the appropriate voice/fax channel as in step 8. You will find that the setting is similar to that established in step 8.

18. Move to the voice/fax Output Level Attenuation configuration menu for the E&M channel No.1 as in step 9.

19. Read the value of the receive level on the No.1 Voice Network Analyzer display. The receive level should fall between –8 and –10 dBm. Set the output attenuation until the receive level display is –8 dBm (e.g., the output attenuation will be between 0 and 2 dB). If the receive level is below (more negative than) –10 dBm, reduce the output attenuation until no further adjustment is possible (0 attenuation). You will find that the setting is similar to that established in step 10.

    Now that both channels are adjusted, test the voice quality. To do that, place both Voice Network Analyzer SETUP switches to BRDG, take one telephone off-hook, dial access code 80 and then the extension number of the other phone. Talk to verify that there are no echoes. If there are echoes, the output attenuation of the distant port is too low. You must increase the output attenuation setting of the distant port.

20. Clear the connection, and restore both channels to System Controlled Busy. If voice switching is to be used, place a switched call to verify that the settings are complete.

21. Remove any and all test equipment, such as Voice Network Analyzers and telephone sets, and restore the system to normal service.

PBX CO Trunk Application Adjustments

If your application involves CO trunks connecting to voice channels, the voice channels will be strapped for FXS rather than E&M type interfaces. In all other respects the adjustment procedure is the same as that given for the PBX Tie Trunk. Use Figure B-4 for test setup details.

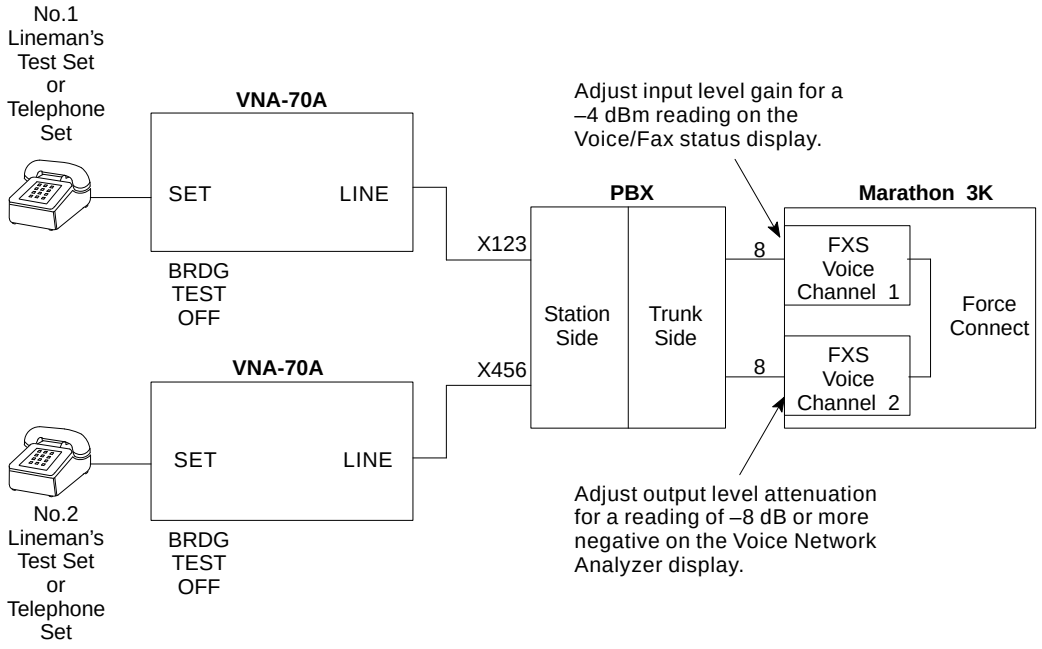


Figure B-4. PBX CO Trunk Application Test Setup

PBX Station or CENTREX PSTN Line Application Adjustments

For this procedure, one voice/fax channel with an FXS interface is required as a companion channel. If neither channel is strapped for FXS, then one voice/fax channel will have to be converted temporarily for an FXS interface. If there are no other voice/fax channels available locally, then the local FXO voice/fax channel will have to be force connected to a companion FXS voice/fax channel across the network. In that case, two technicians will be required to perform the adjustment procedure, one at each end.

Note: If there are no single-line telephone circuits available from the PBX or CENTREX/PSTN, then the line assigned to the channel that is temporarily converted to FXS can be used for test purposes.

1. Use the test setup shown in Figure B-5. Initially, set both Voice Network Analyzers to BRDG, TEST, and OFF. The extension numbers in Figure B-5 are fictitious, used here for reference only.

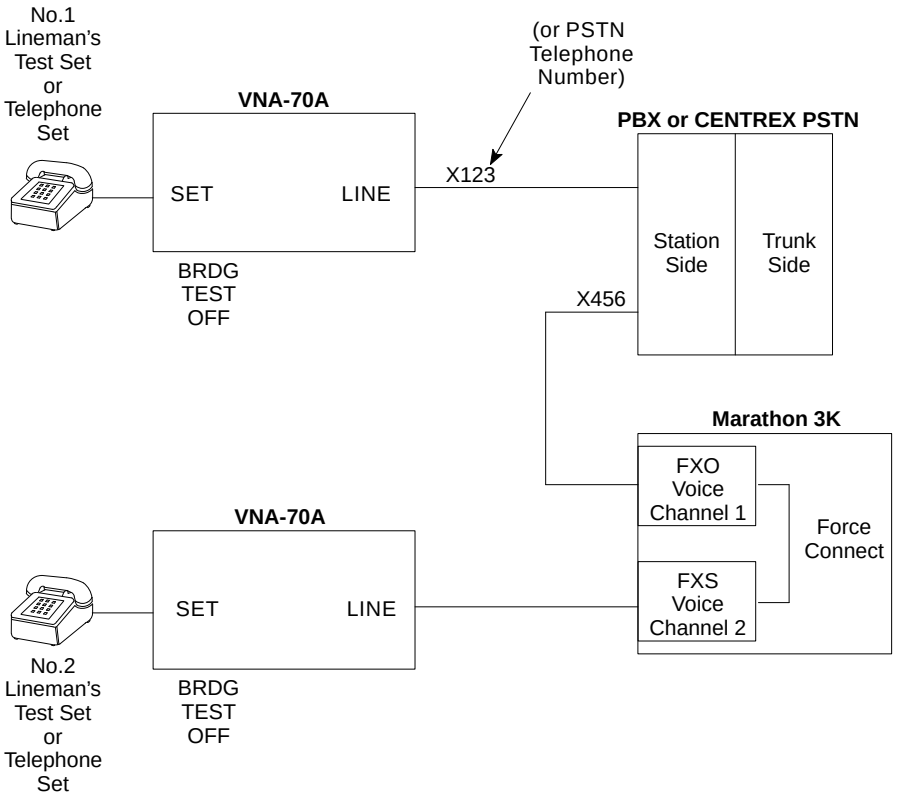


Figure B-5. PBX Station/Centrex Line Application Test Setup

2. Force connect the FXO channel to the companion FXS channel.
3. Place a call from PBX extension 123 to extension 456. To do that, lift the No. 1 telephone off-hook, wait for the dial tone from the PBX, then dial extension number 456. The No. 2 telephone set will ring. (In CENTREX PSTN applications, dial the published PSTN number).
4. Place the SETUP switch on the No.2 Voice Network Analyzer to TERM. The telephone will stop ringing.
5. Place the SETUP switch on the No.1 Voice Network Analyzer to TONE, and disconnect the No.1 telephone from the Voice Network Analyzer.

6. Using the asynchronous terminal, enter the Voice/Fax Status. To do that, follow this path:

   Command Facility Main Menu

   Status/Statistics

   ↓

   Voice/Fax Status

   ↓

   ENTER NODE-ID/CHANNEL # (OR RANGE OF CHANNELS) (^X TO ABORT):

7. Observe the display on the asynchronous terminal, determine which channel is receiving a tone, and note its input level. If the reading is more positive than –4 dBm, apply attenuation (negative gain) in the amount that this reading is above –4 dBm. If the reading is more negative than –4 dBm, add gain (6 dB maximum) in the amount that the reading is below –4 dBm. (Normally, station circuits will require an input level gain setting of between +2 and +4 dB in order to obtain the –4 dBm reading. Most likely you will not have to attenuate the input level. CENTREX or PSTN lines will normally require +2 to +6 dB input level gain.

   To adjust the input level for the FXO channel, enter the Input Level Gain Menu for the appropriate voice/fax channel, by following these steps:

   Command Facility Main Menu

   Configure Local Nodes

   ↓

   Voice/Fax Menu

   ↓

   Voice/Fax Channel Configuration

   ↓

   ENTER NODE-ID/CHANNEL # (^X TO ABORT):

   ↓

   Voice/Fax Channel Characteristics

   ↓

   Input Level Gain

   INPUT LEVEL GAIN [0 dB]

   After the input level gain adjustment is completed, jot down the setting used to bring the input level to –4 dBm. You will need it for comparison when setting other FXO channels.

8. Read the value of the receive level on the No.2 Voice Network Analyzer display. Jot down this value for comparison when adjusting other FXO channels.

   You have now established the input level gain setting on the FXO voice channel. In the following steps you will adjust the output level attenuation setting on the FXO channel.

9. Reconnect the No.1 telephone set to its Voice Network Analyzer.
10. Set the SETUP switch on both Voice Network Analyzers to BRDG.
11. Take the No.2 telephone set off-hook. You will receive a dial tone from the PBX across the force-connected voice channels. Dial extension 123 (or PSTN telephone number). The No.1 telephone set will ring.
12. Place the SETUP switch on the No.1 Voice Network Analyzer to TERM. The telephone will stop ringing.
13. Place the SETUP switch on the No.2 Voice Network Analyzer to TONE, and disconnect the No.2 telephone from the Voice Network Analyzer.
14. Using the asynchronous terminal, enter the Voice/Fax Channel Status as in step 6.
15. Observe the display on the asynchronous terminal and note the input level of the FXS channel. If the reading is more positive than –4 dBm, apply attenuation (negative gain) in the amount that this reading is above –4 dBm. If the reading is more negative than –4 dBm, add gain (6 dB maximum) in the amount that the reading is below –4 dBm. To adjust the input level, enter the Input Level Gain Menu for the appropriate voice/fax channel as in step 7.
16. Read the value of the receive level on the No. 1 Voice Network Analyzer display. The receive level should be –12 dBm or lower. In this case there are no adjustments to be made on the output level attenuation, since the 0 dB setting (no attenuation) is the maximum obtainable level. Note down this level for comparison when adjusting other FXO channels.
17. Now that both channels are adjusted, test the voice quality. To do that, place both Voice Network Analyzer SETUP switches to BRDG, take the No. 2 telephone off-hook, wait for the dial tone and dial extension 123. Talk to verify that there are no echoes. If there are echoes, the output level attenuation of the No.1 channel is too low. You must increase the output attenuation setting of the FXO channel.
18. Restore the FXS voice channel to FXO; it was used temporarily as a companion device with the FXO voice channel under test. Clear all local force connects and restore all channels to System Controlled Busy. If voice switching is being used, place a switched voice call to verify that the adjustments are correct.

Note: Local switched calls cannot be accomplished unless the FXO interface is set for POWER DISCONNECT option. Remove any test equipment installed and restore the system to normal service.

Single Line Telephone Application

1. Use the test setup shown in Figure B-6. Initially, set both Voice Network Analyzers to BRDG, TEST, and OFF.

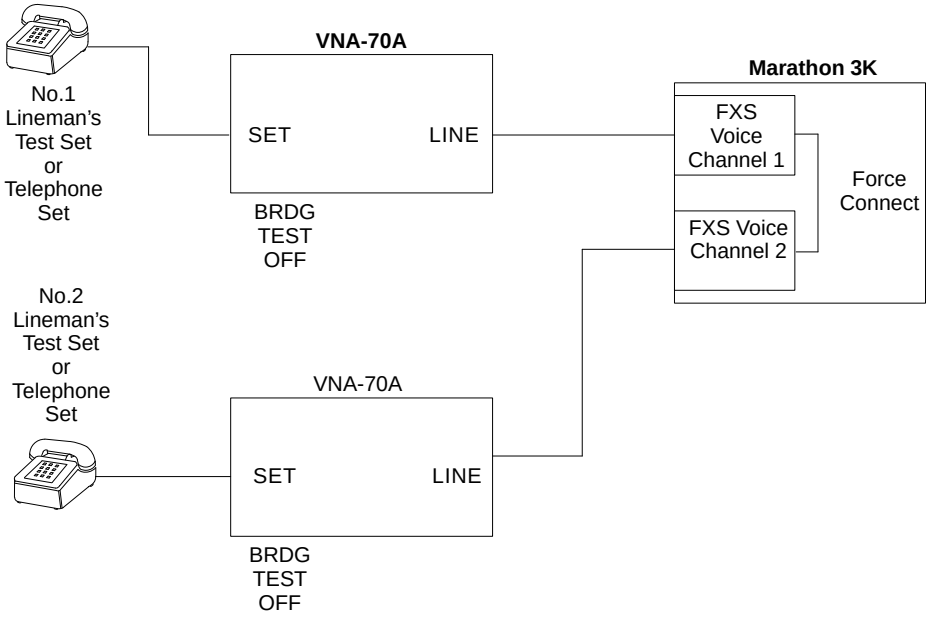


Figure B-6. Single Line Telephone Test Setup

2. Force connect a pair of FXS channels in the Integration unit.
3. Lift the No.1 telephone set off-hook. The No.2 telephone set will ring.
4. Place the SETUP switch on the No.2 Voice Network Analyzer to TERM. The telephone will stop ringing.
5. Place the SETUP switch on the No.1 Voice Network Analyzer to TONE, and disconnect the No.1 telephone from the Voice Network Analyzer.
6. Using the asynchronous terminal, enter the Voice/Fax Status. To do that, follow this path:

Command Facility Main Menu

↓ Status/Statistics

↓ Voice/Fax Status

↓ ENTER NODE-ID/CHANNEL # (OR RANGE OF CHANNELS) (^X TO ABORT):.

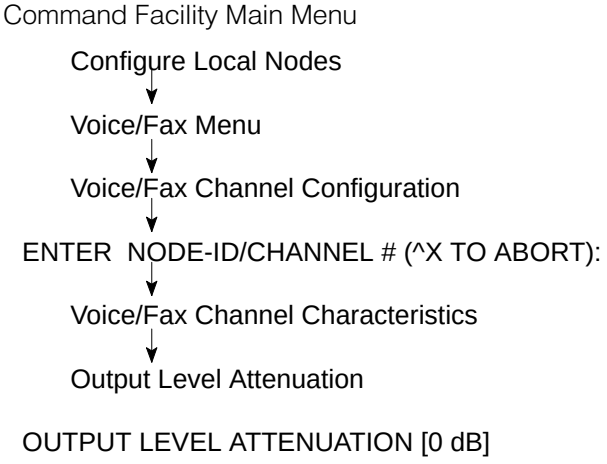
7. Observe the display on the asynchronous terminal, determine which channel is receiving a tone, and note its input level. If the reading is more positive than –4 dBm, apply attenuation (negative gain) in the amount that this reading is above –4 dBm.

   Normally, this will require the Input Level Gain to be set between –2 and –4 dBm depending on cabling loss. (Note: if the Input Gain is currently set for 0 and the Input Level is below –10 dBm, then the FXS channel should be evaluated for a problem.)

   To adjust the input level for the No.1 FXS channel, enter the Input Level Gain Menu for the appropriate voice/fax channel as in step 7 of the PBX Station Application procedure.

8. After the input level gain adjustment is completed, note down the setting used to bring the input level to –4 dBm. You will need it for comparison when setting the other FXS channels.

9. Move to the voice/fax Output Level Attenuation configuration menu for the FXS channel No.2 by following these steps:



10. Read the value of the receive level on the No.2 Voice Network Analyzer display. The receive level should fall between –6 and –8 dBm. Set the output attenuation until the receive level display is –8 dBm. If the receive level is below (more negative than) –8 dBm, reduce the output attenuation until no further adjustment is possible (0 attenuation). Note down the setting. You will need it for comparison when the setting other FXS channel.

11. You have now established the input level gain setting on the No.1 channel, and the output level attenuation setting on the No.2 channel. Next, adjust the input level gain setting on the No.2 channel, and the output level attenuation setting on the No.1 channel.

12. Reconnect the No.1 telephone set to its Voice Network Analyzer.

13. Set the SETUP switch on both Voice Network Analyzers to BRDG.
14. Take the No.2 telephone set off-hook. The No.1 telephone set will ring.
15. Place the SETUP switch on the No.1 Voice Network Analyzer to TERM. The telephone will stop ringing.
16. Place the SETUP switch on the No.2 Voice Network Analyzer to TONE, and disconnect the No.2 telephone from the Voice Network Analyzer.
17. Using the asynchronous terminal, enter the Voice/Fax Status as in step 6.
18. Observe the display on the asynchronous terminal, determine which channel is receiving a tone, and note its input level. If the reading is more positive than –4 dBm, apply attenuation (negative gain) in the amount that this reading is more positive than –4 dBm.

    To adjust the input level, follow the procedure of step 7 above. You will find that the setting is similar to that established in step 8.
19. Move to the voice/fax Output Level Attenuation configuration menu for the FXS channel No.1 as in step 9 above.
20. Read the value of the receive level on the No.1 Voice Network Analyzer display. The receive level should fall between –6 and –8 dBm. Set the output attenuation until the receive level display is –8 dBm. If the receive level is below (more negative than) –8 dBm, reduce the output attenuation until no further adjustment is possible (0 attenuation).
21. Now that both channels are adjusted, test the voice quality. To do that, place both Voice Network Analyzer SETUP switches to BRDG, take one telephone off-hook and lift the other telephone off-hook when it rings. Talk to verify that there are no echoes. If there are echoes, increase the output level attenuation setting on the distant channel.
22. Clear all force connections, and restore both channels to System Controlled Busy. If voice switching is to used, place a switched call to verify that the settings are complete.
23. Remove any and all test equipment, such as Voice Network Analyzers and telephone sets, and restore the system to normal service.

Hybrid Key System Adjustment

If your installation includes a hybrid key system instead of a single-line telephone, there may be a loss of up to 3 dB across that system in each direction. To compensate for that loss in any application, you will have to adjust the input/output levels accordingly.

In the test setup shown in Figure B-7, you will need only 1 dB attenuation on voice channel No.1 for an input level value of –4 dBm. Similarly, when adjusting the output level on voice channel No.1, you will most likely require little or no attenuation in order to maintain the output level between –8 and –10 dBm. In all other respects, this adjustment procedure is the same as the single-line telephone adjustment procedure described in the previous paragraph.

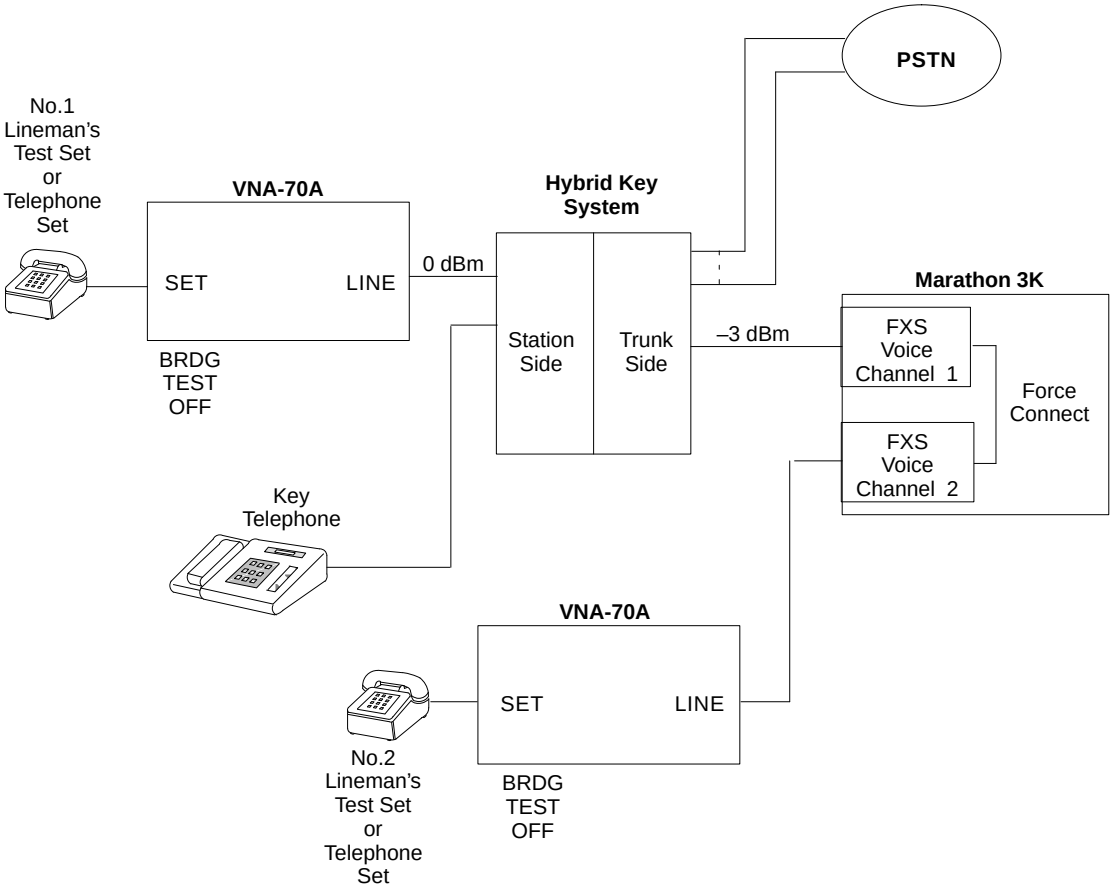
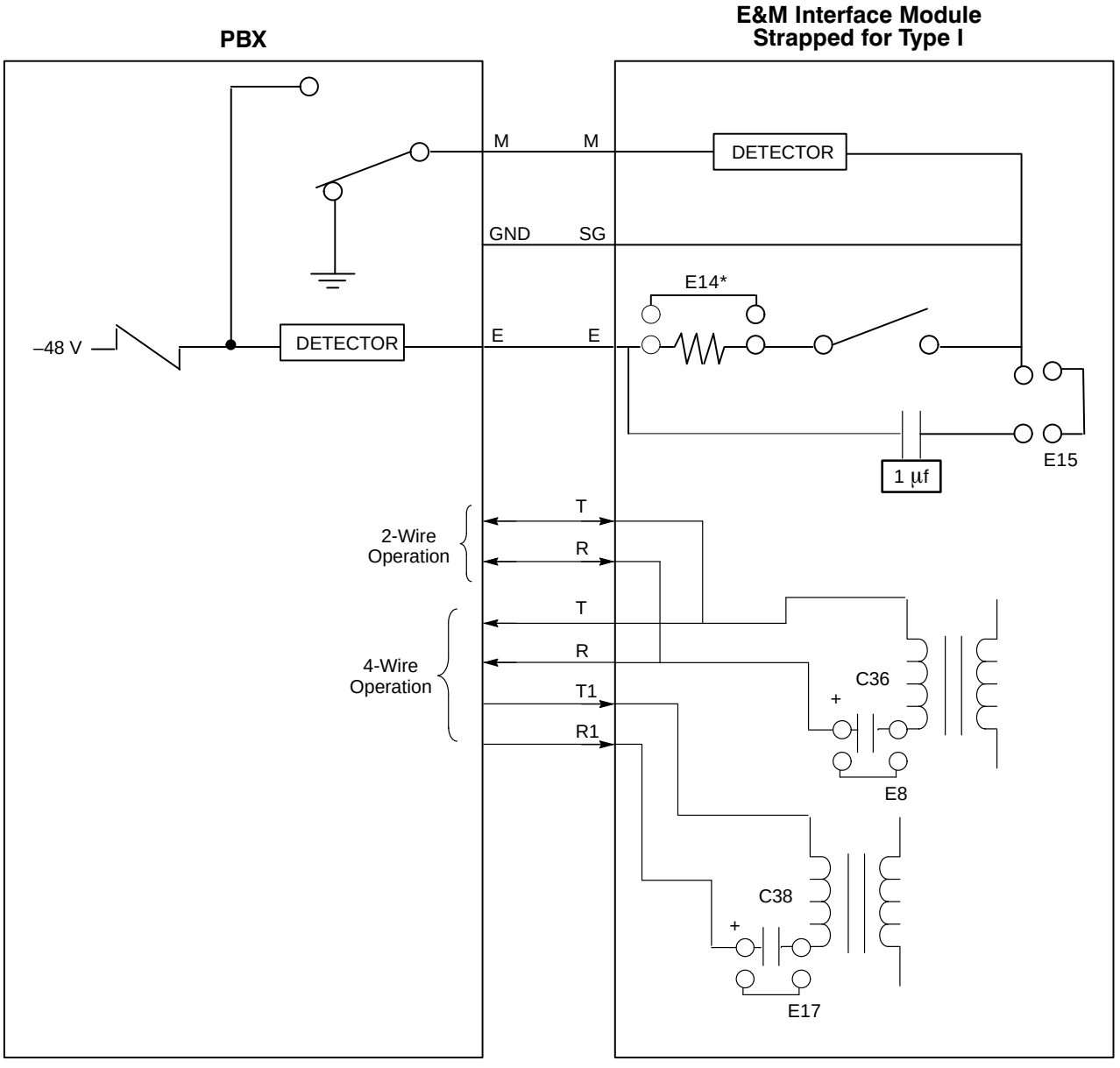


Figure B-7. Hybrid Key System Test Setup

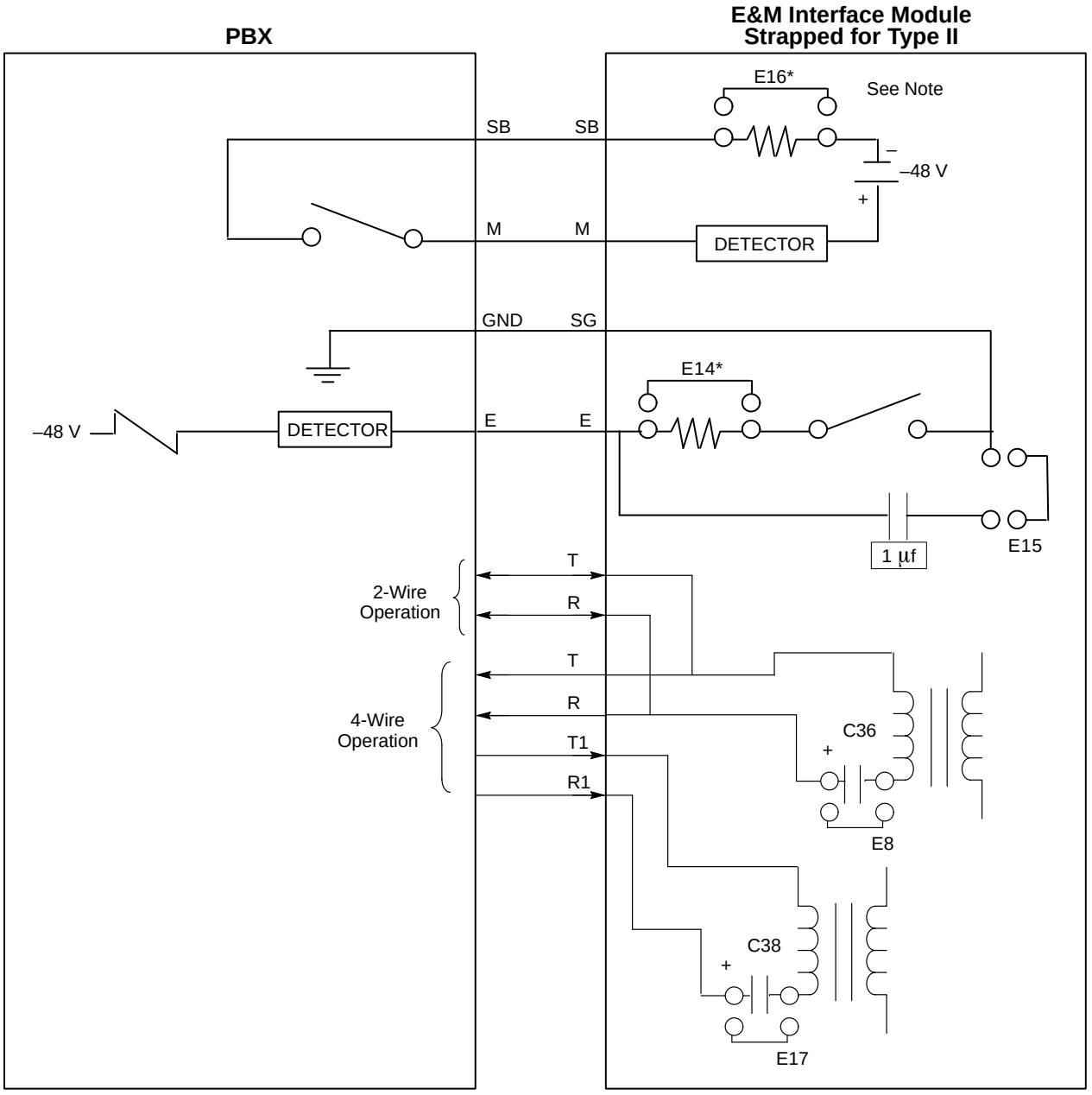
PBX Interface Connection Diagrams C

Figures C-1 through C-3 show the circuit connections between the E&M-strapped voice/fax channel and the PBX for Signalling Types I, II, and V.



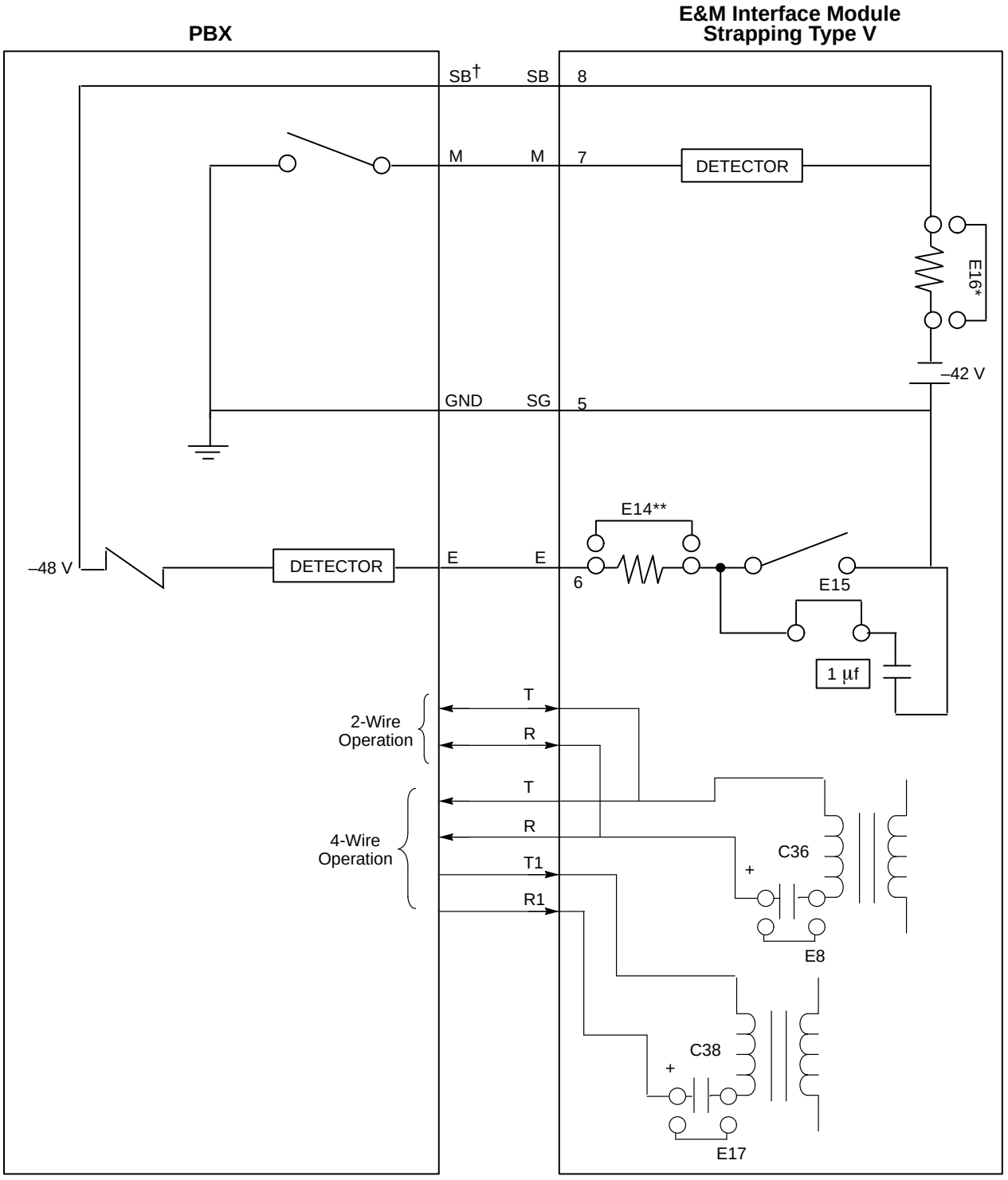
*E-14 removed if long distance cables require it.

Figure C-1. Type I E&M Signalling Interface Connection Diagram



Note: It is recommended that strap E16 and E14 be removed if cable distances require it.

Figure C-2. Type II E&M Signalling Interface Connection Diagram



* Shipped from the factory with E16 not installed. Install only when SB lead loop resistance is too high.
** Shipped from the factory with E14 not installed. Install only when E lead loop resistance is too high.
† CAUTION: PBX SB and interface module SB are to be connected only when –48 volt supply is not available on the PBX side.

Figure C-3. E&M Type V Interface Connection Diagram

Fax and Modem Operation D

Fax Operation

The voice/fax channel can accept analog Group 3 fax signals at 9600 bps, 7200 bps, 4800 bps, or 2400 bps for transmission to the remote end. The fax capability is implemented by firmware resident within the voice/fax channel. The firmware constantly monitors the incoming analog stream for the presence of fax signals. When fax signals are detected, the firmware switches the voice/fax channel into the fax mode. While in this mode, the signals are demodulated into fax symbols and passed as data to the remote end. At the remote end the signals are remodulated to the correct standards. All fax/voice operations within the voice/fax module are automatic and transparent to the user.

Group 3 signals are supported at rates equal to or less than the digitization rate of the voice/fax channel. Specifically, voice/fax channels configured at 9600 bps or greater will support fax signals at 9600 bps (or at lower rates, depending on machine compatibility). The fax signal will only use link bandwidth at the operating speed in the direction of transmission, independent of the digitization rate configured in the voice/fax channel. For example, if the digitization rate of the voice/fax channel is 9600 bps and the fax signal rate is 7200 bps, the fax signal will only use up 7200 bps (plus overhead) of the link bandwidth.

Fax Machine Types

Since there are numerous brands and types of fax machines currently available to the user. An example of a suitable fax machine is the Ricoh Model 2100 or the Pitney Bowes Model 8230. In general, any model Group 3 fax machine with auto-answer capability will work with the voice/fax module.

Auxiliary Devices

There are two auxiliary devices associated with an FXS-strapped voice/fax channel: a line sharing unit, and a fax sharing unit.

Line Sharing Unit

This device is used in applications where fax and voice share the same voice/fax channel. Its main function is to monitor the inbound analog signal obtained from the voice/fax channel, and to route this signal either to the fax machine or telephone set, as appropriate. Outbound signals are routed by the unit transparently from either the telephone set or fax machine to the voice/fax channel, for ultimate transfer to the remote end.

Figure D-1 shows how the Line Sharing Unit connects to the different interfaces at one end of the voice/fax channel. There are three ports on the unit: common port; voice port; and fax port. The common port connection is a telephone cable with a male RJ11 jack that plugs into the RJ11 connector at the rear of the 3K Base Module. The voice port is a female RJ11 connector that connects to the telephone set, while the fax port is a female RJ11 connector that connects to the fax machine.

Operating power to the Line Sharing Unit is obtained from an ac outlet through the ac adapter. In case of power failure the unit defaults to the voice mode, effectively disconnecting the fax machine.

Note that the number of ports, connector labels, and power connections shown in Figure D-1 are typical, and may vary from one device to the next. Some line sharing models may include an extra port for an answering machine, while others may feature special functions not described here.

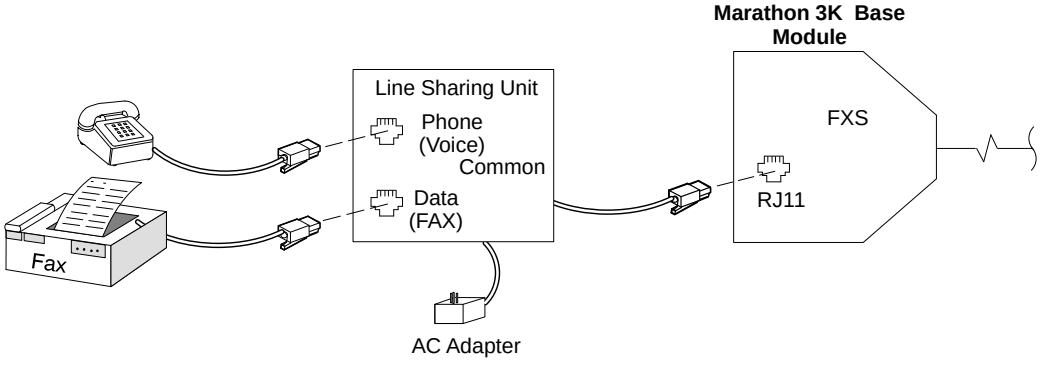


Figure D-1. Line Sharing Unit Connection Diagram

Fax Sharing Unit

This device is used when the fax machine is shared between the Public Switched Telephone Network (PSTN) and the Integration network. It is a semi automatic device; automatic for inbound calls, dial select for outbound calls.

Figure D-2 shows how the Fax Sharing Unit connects to the different interfaces at one end of the voice/fax module. There are three ports on the unit labelled FAX, PORT 1, and PORT 2, each terminated in a female RJ11 connector. Connection to the unit is made by modular cord, with the fax machine connecting to the FAX port. Of the two line ports, one connects to the PSTN and the other to the voice/fax channel RJ11 connector..

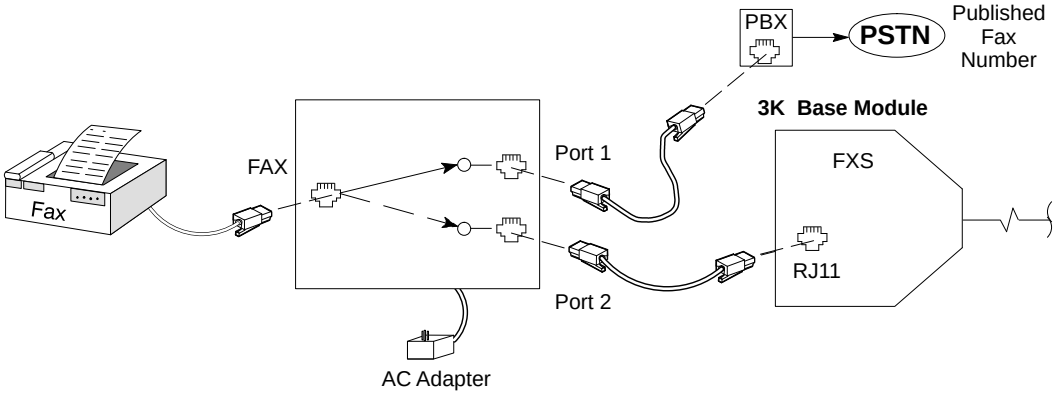


Figure D-2. Fax Sharing Unit Connection Diagram

Fax Applications, Force-Connected Channels

Transmitting fax over the Integration network involves many factors, such as: your current type of data/voice installation; whether or not you are already connected to the PSTN network and have a published fax number; whether you plan to use the same channel for fax and voice or have a channel dedicated to fax only; etc.

In the following paragraphs a number of applications are shown and discussed. For clarity, equipment involved in data traffic is omitted from these discussions.

FXS-to-FXS Shared Channel

Figure D-3 shows an installation involving a headquarters office in Los Angeles connected to a field office in Simi Valley and anchored by a 3K Base Module at each end. Each voice/fax channel is strapped for FXS, and used for both voice and fax traffic. To that end, a Line Sharing Unit is installed at each end to merge the fax machine and telephone set to the voice/fax channel.

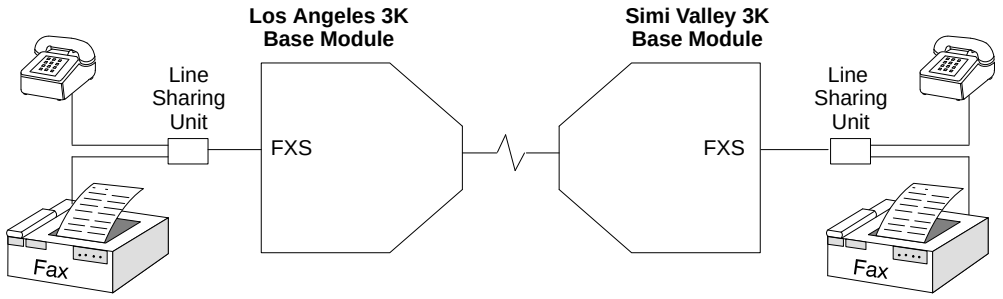


Figure D-3. FXS-to-FXS Fax Application

To Send a Fax

Using the hookup shown in Figure D-3, proceed as follows:

Note: It is assumed here that both fax machines are powered up, and programmed for auto answer.

1. At the end from where you intend to initiate the fax message insert the paper, dial any number, and press the START (or SEND) push-button on the fax machine.
2. All operations will now follow automatically. Paper will be feeding and the OK indicator associated with the voice/fax channel will flash.
3. At the completion of the fax transmission the OK indicator will stop flashing, and the voice/fax channel will return to the voice mode.

To Make a Telephone Call

If the fax machine is not in use, you may make a telephone call at any time by simply lifting the receiver on the telephone set. It will ring at the other end, and stop ringing when the other party answers.

Note: If the fax machine is already in use, you will get a busy signal.

PBX-to-PBX Shared Channel/Dedicated Fax

Figure D-4 shows an application for a shared fax/telephone hookup over the Integration network. Each end has a 3K Base Module, voice PBX, fax machine, and telephone set. The fax machines are dedicated for internal company use and share the voice/fax channel with the voice traffic. Both fax machines and telephone sets are connected to the station side of the PBXs, while the 3K Base Module are connected to the trunk side. The fax machines were assigned station numbers 2000 and 4000, respectively, while the telephone sets were assigned station numbers 1234 and 4567, respectively. Similarly, numbers 81 and 82 were assigned to the trunks at each end. All numbers used in this example are arbitrary; you may choose any numbers that suit your installation. Each voice/fax channel is strapped for E&M Type I.

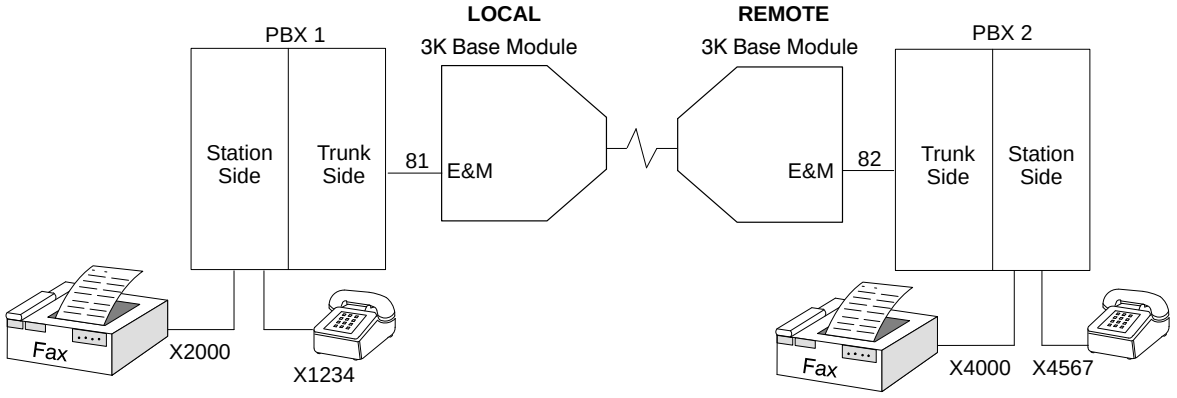


Figure D-4. PBX-to-PBX Shared Channel/Dedicated Fax Application

To Send a Fax

Using the hookup and PBX port assignments shown in Figure D-4, proceed as follows:

Note: It is assumed here that both fax machines are powered up, and programmed for AUTO ANSWER.

1. At the end you intend to initiate the fax message, insert the paper, dial the trunk number, and wait for the dial tone from the far end PBX.
2. Dial the station extension number, and listen for the answer from the other end. This will be a 2100-Hz tone followed by a variable-frequency tone.
3. Press the START (or SEND) push-button.
4. All operations will now follow automatically. The paper will be feeding and the OK indicator on the voice/fax module channel will flash.
5. At the completion of the fax transmission the OK indicator will stop flashing, and the voice/fax channel will return to the voice mode.

To Make a Telephone Call (per format in Figure D-4)

Voice operations in the example shown in Figure D-4 are the same as voice operations in any installation that does not include fax, as long as the voice/fax channel is not busy with fax operations.

1. Lift the receiver at 1234, wait for the dial tone, and dial 81.
2. Wait for the second dial tone, then dial extension number 4567.
3. Wait until the other party answers, then start your conversation.

PBX-to-PBX Shared Channel/Shared Fax

An expanded version of the application shown in Figure D-4, is illustrated in Figure D-5. Here, the fax machines not only share the voice/fax channel with voice traffic, but may be used alternately between the Integration network and the external PSTN. To that end, each fax machine has a published 7-digit fax number, and is accessible to any user connected to the PSTN.

To allow outside callers to reach the fax machine without having to pass through an attendant, the PBX system must be able to support Direct Inward Dialing (DID) or Direct In Termination (DIT). DID allows an outside caller to reach an internal extension without going through an attendant or operator. DIT routes a call made to the published fax number directly to the fax machine.

To send a fax or make a telephone call over the Integration network, follow the same procedure given for the shared channel/dedicated fax application (page D-5). To send a fax to a published number over the PSTN, dial first the outside trunk access number (typically 9), wait for the second dial tone, then dial the published trunk number. This allows use of least cost routing features in the PBX. Alternatively, a unique trunk access number other than 9 may be assigned to the fax machine, so that it would be automatically busy in the PSTN when the fax machine is sending on the PSTN.

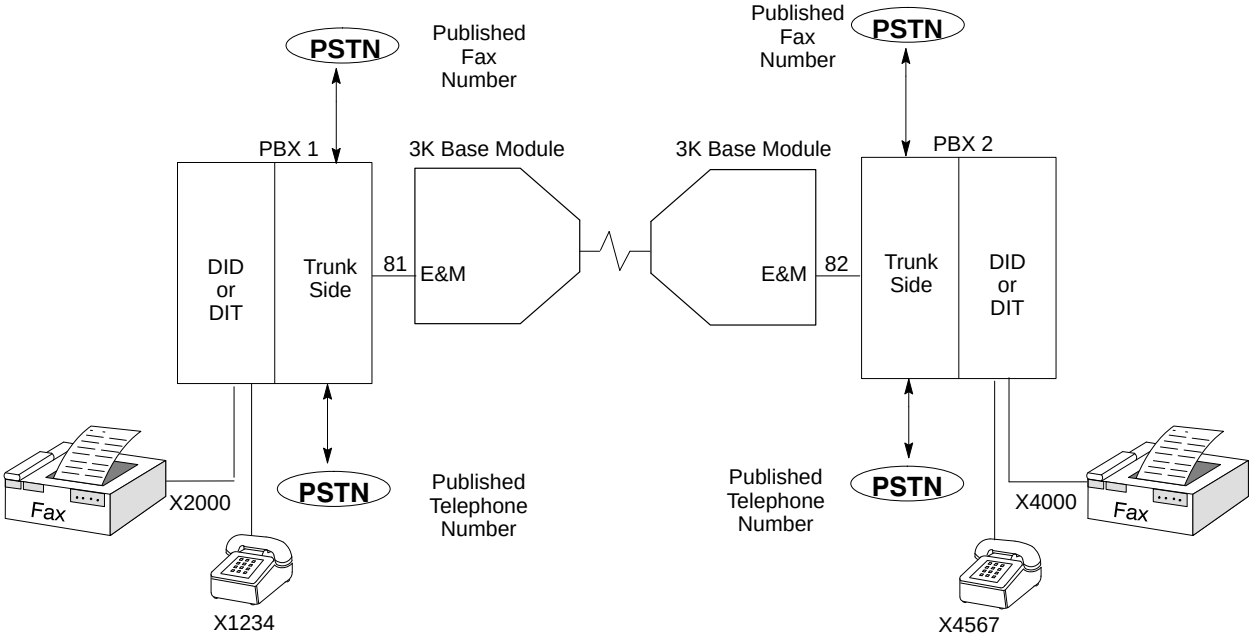


Figure D-5. PBX-to-PBX Shared Channel/Shared Fax Application

PBX-to-Hybrid or KTS Shared Channel with Dedicated Fax

In this application, Figure D-6, PBX station number 789 is extended over the Integration network to a remote key system. At the FXS end, a Line Sharing Unit allows alternate use of the voice/fax channel between a dedicated fax machine and a telephone set connected to the key system. The voice/fax channel at the PBX end is strapped for FXO, while the voice/fax channel at the key system end is strapped for FXS.

To send a fax from the PBX end to the dedicated fax machine, the fax machine user at station number 456 dials 789, and the Line Sharing Unit will automatically route the call to the fax machine. Similarly, to make a telephone call from the PBX end to the telephone set at the FXS end, the telephone user at station 123 dials 789, and the call is automatically routed by the Line Sharing Unit to the telephone set.

Note that both the telephone and fax machine stations at the PBX can access (or be reached from) the PSTN network. Similarly, the telephone set at the FXS can access (or be reached from) the PSTN network.

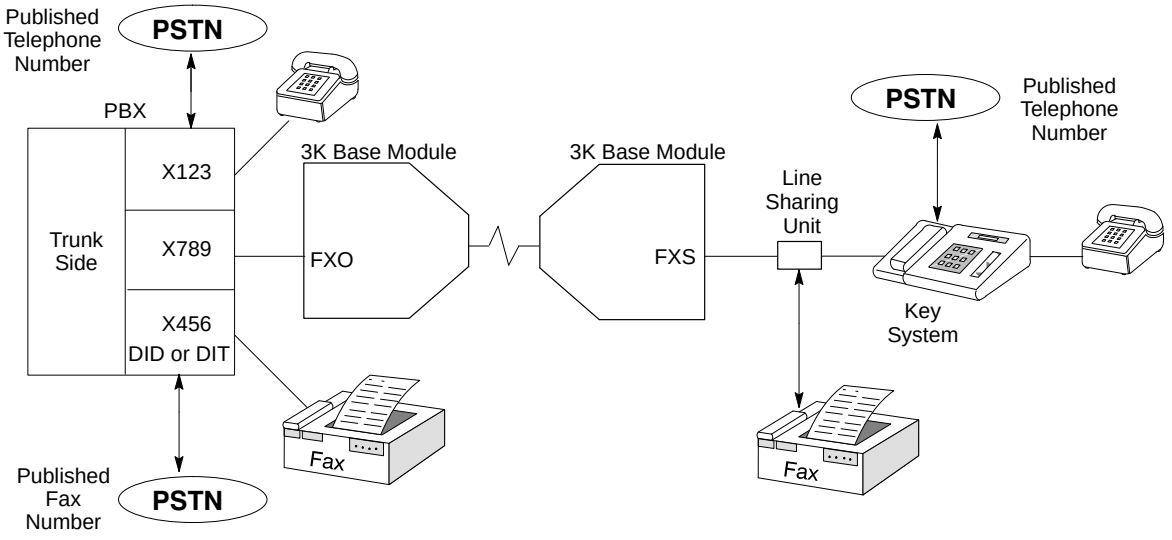


Figure D-6. PBX-to-Hybrid or KTS Shared Channel with Dedicated Fax Application

PBX-to-KTS Shared Fax with Shared Channel

This application, Figure D-7, is similar to the Shared Channel/Dedicated Fax application illustrated in Figure D-6. Comparing Figure D-7 with Figure D-6, it can be seen that in Figure D-6 the fax machine at the FXS end has a published fax number and is connected to a Fax Sharing Unit. Accordingly, the fax machine is shared between the Integration network and the PSTN. And, as in the previous case, the Integration voice/fax channel is shared between the fax machine and the key system, made possible by the Line Sharing Unit.

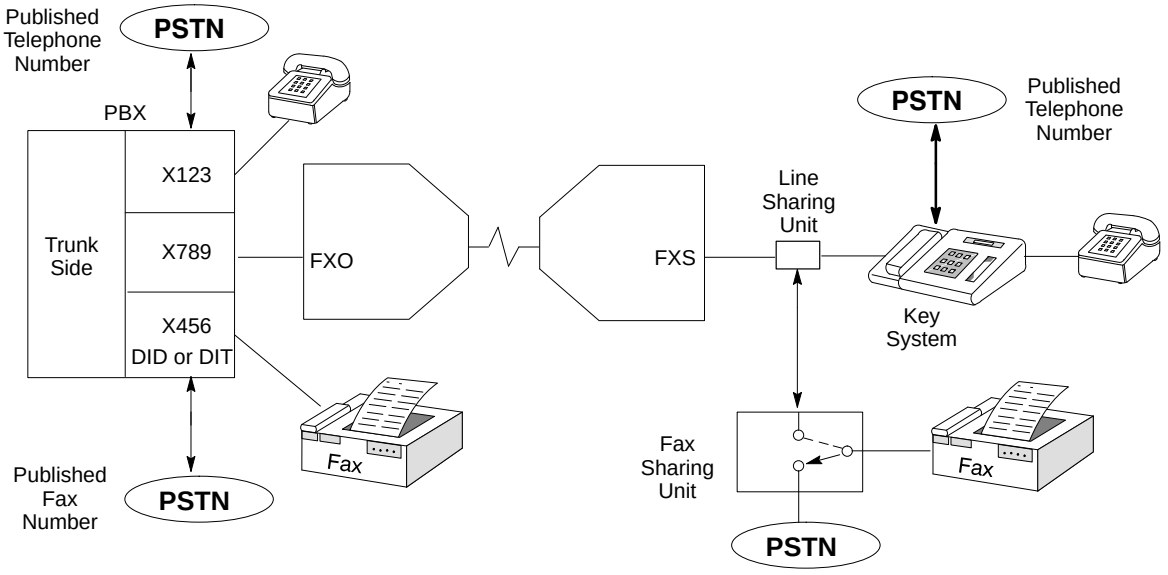


Figure D-7. PBX-to-KTS, Shared Fax with Shared Channel

Modem Operation

In addition to voice and fax, the voice/fax channel can process data obtained from a modem for transmission across the link.

Models compatible with the voice/fax module include V.22 and V.22 bis modems.

Modem demodulation, like fax demodulation and voice compression, is implemented by firmware resident within the voice/fax channel. The firmware can differentiate between signals obtained from a modem, fax machine, or telephone. When the presence of modem signals is detected, the firmware switches the voice/fax channel into the modem mode. While in this mode, the signals are processed accordingly and passed as data to the remote end. At the remote end the signals are remodulated to modem data.

Network Code Download E

Each voice/fax channel is equipped with Flash memory that allows local or remote updating of operating software. The downloaded files are sent to their destination Flash memory devices in an asynchronous data stream.

This Appendix contains procedures for downloading voice/fax channel software into Flash memory.

Methods of Downloading

- Local or remote, using BLACK BOX's PC-based terminal emulation software in a WINDOWS environment.
- Local or remote, using BLACK BOX's PC-based terminal emulation software in a DOS environment.
- Local or remote, running NETMAN.

The methods described in the following paragraphs involve a local PC, using BLACK BOX's terminal emulation software in both WINDOWS and DOS environments.

Prerequisites

An IBM-Compatible, Intel 286-based (or better) PC with

- A high-density 3 ½-inch floppy drive.
- A hard drive.
- An asynchronous COM port on the PC.

A copy of FlashDLD terminal emulation is included in the diskette labelled **UVM Flash Code Download Software**. This diskette is shipped with each Marathon 3K unit.

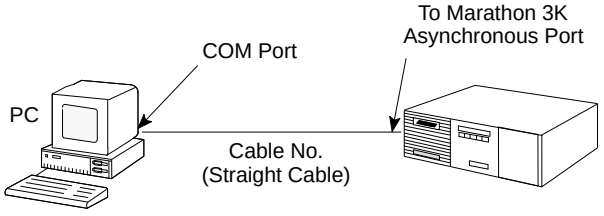
Procedures for Code Download Using a PC

Summary

Connection
Installing the FlashDLD Program • DOS • WINDOWS
Starting the Communications Software
Configuring the Communications Software • DOS • WINDOWS
Configuring the Download Parameters • Log into the Command Facility • Set the Password • Set the Inactivity Timeout
Erasing and Exit • Prerequisite for ClearVoice Technology • Erase the Flash Image • Exit the Command Facility
Initiating the Code Download • Get the ENTER CLASS prompt • Enter the file transfer mode – DOS – WINDOWS

Connection

The PC must be connected to the Marathon 3K unit. To do that, connect an RS-232 straight cable between COM1 or COM2, as appropriate, and one of the asynchronous ports on the Marathon 3K as shown below.



Installing the FlashDLD Program

BLACK BOX's terminal emulation software FlashDLD was designed to download code specifically to BLACK BOX products. It is available in two versions: one for DOS and one for WINDOWS. Both versions are included on the FlashDLD diskette. Perform either of the following procedures as applicable to your installations.

FlashDLD Diskette

FlashDLD for DOS

Note: This simplified installation procedure will work with most PCs. It is assumed that the 3 ½-inch diskette drive is in drive A, and that the software is to be installed on drive C in the default directory named FLASHDLD.

1. Place the diskette labelled **UVM Flash Code Download Software** into diskette drive A (or B, if applicable) of your PC.
2. Log on to the diskette drive and change directories to the FlashDLD directory:

```
c:\>a:
a:\> cd flashdld
a:\flashdld>
```

3. Enter the dossetup command:

```
a:\flashdld> dossetup
```

   The command syntax is :dossetup {source destination}. The default destination is C:\FLASHDLD.

4. As dossetup starts up, an Option Selection screen will be displayed. Select both FlashDLD Executable and Flash Download Files, continue with the installation.

The FlashDLD for DOS program is now installed on your PC.

FlashDLD for WINDOWS

Note: This simplified installation procedure will work with most PCs. It is assumed that the 3½" diskette drive is in drive A, and that the software is to be installed on drive C in the default directory named FLASHDLD. For special requirements refer to the Code Download User's Manual, 800-1844 for more detailed instructions.

1. Place the diskette labelled **UVM Flash Code Download Software** into the diskette drive of the PC.
2. From the Program Manager, select **File** → **Run**. In the Command Line box, type the following:

   ```
   a:\flashdld\winsetup.exe
   ```

   Then click on OK.
3. When the Setup screen appears, click on **Continue** to proceed with the software installation.
4. In the Destination Path screen, click on **Continue** to select the default software installation directory of c:\flashdld.
5. In the Option Selection screen, select both FlashDLD Executables and Flash Download Files, then click on **Continue**.
6. In the Destination Group screen, click on **Continue** to create the default group called FlashDLD.

The FlashDLD for WINDOWS program is now installed on your PC.

Starting the Communications Software

For FlashDLD for DOS, at the DOS prompt enter the following command:

```
c:\flashdld> flashdos
```

For FlashDLD for WINDOWS, double click on the FlashDLD for WINDOWS icon.

Configuring the Communications Software

Set the communications software to the following parameters:

```
9600 baud
No parity
8 data bits
1 stop bit
```

For FlashDLD for DOS, use **Alt-S** to enter the setup mode and **Esc** to exit. When prompted to "Save this configuration?", answer Y.

For FlashDLD for Windows, select **Setup** on the menu to enter the setup mode.

Once the communications software has been configured for the code download, set the software to the terminal emulation mode. This is done automatically in FlashDLD, after the configuration has been saved.

Configuring the Download Parameters

1. **Log into the Command Facility**

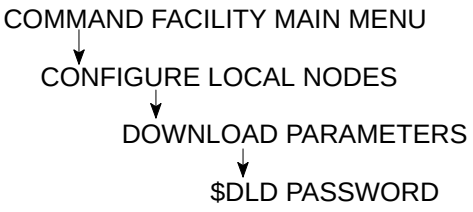
   a. Press the Enter key to get the unit's ENTER CLASS prompt:

   ```
   ENTER CLASS:
   ```

   b. Connect to the Command Facility (class $CMD).

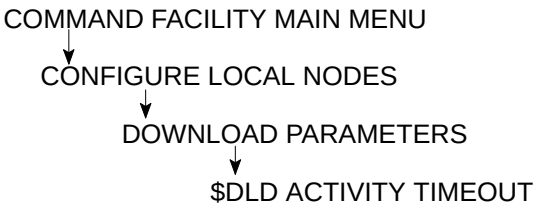
   ```
   ENTER CLASS: $cmd
   ENTER CLASS PASSWORD
   ```

2. **Set the password:**



Enter the password for the code download facility ($DLD). The password can have a maximum of 8 characters out of the set of A through Z and 0 through 9. The password is not case-sensitive.

3. **Set the Inactivity Timeout**

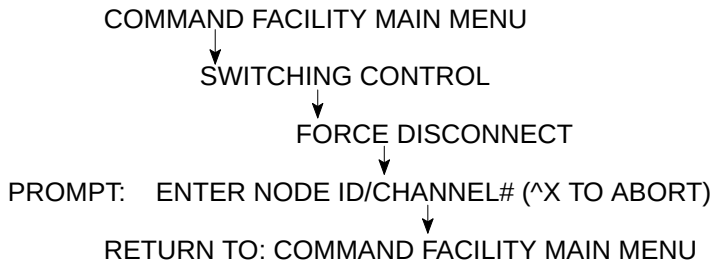


Enter the timeout in seconds. This is the length of time the unit's $DLD facility will wait to disconnect if data transmission has ceased. The default is 120 seconds.

Erasing and Exit

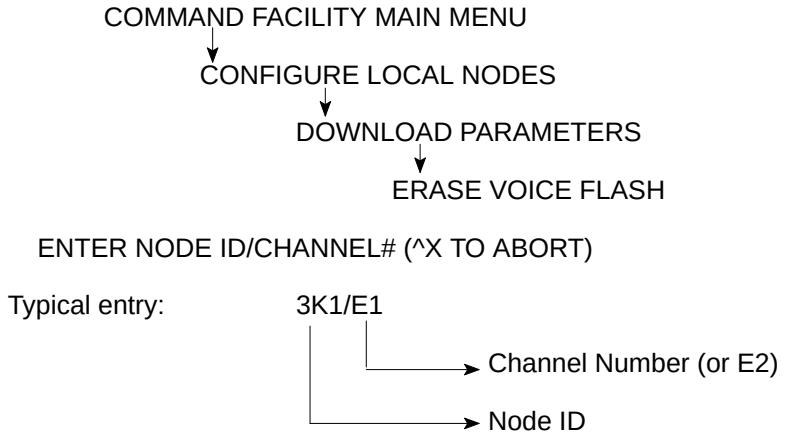
1. Prerequisite for ClearVoice Technology (Force Connect Only)

When attempting to download ClearVoice application code for the first time, before erasing the existing code enter the Command Facility Main Menu and follow this path:


COMMAND FACILITY MAIN MENU
SWITCHING CONTROL
FORCE DISCONNECT
PROMPT: ENTER NODE ID/CHANNEL# (^X TO ABORT)
RETURN TO: COMMAND FACILITY MAIN MENU

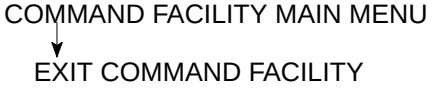
When code download is completed, restore the force connection.

2. Erase the Flash Image



While the voice flash is erasing (approximately 5 to 7 seconds), all indicators associated with the applicable voice/fax channel will go on. At the end, the message ERASE COMPLETE will appear on the screen, the OK indicator will be flashing, and all other indicators will go off.

3. Exit the Command Facility



You will receive the following typical (default) message:

```
DISCONNECTED
```

Initiating the Code Download

1. Press the Enter key to get the unit's ENTER CLASS prompt, and then type:

   ```
   ENTER CLASS: 3K/E1/$DLD
   ENTER PASSWORD:
   ```

   a. Under normal circumstances you will receive the following message:

      ```
      CONNECTED
      ```

   b. If the flash memory is not erased, you will receive the following message:

      ```
      BUSY
      ```

   c. If you entered the wrong password, you will receive the following message:

      ```
      UNASSIGNED
      ```

2. **Enter the file transfer mode. To do that,**

   a. For **DOS**, type **Alt-F**, then select the file to be downloaded.

   b. For **WINDOWS**, select **File → Send**, highlight the file to be downloaded, then click on **OK**.

It will take approximately 3 to 5 minutes to complete the loading operation, depending on the size of the file. While the file is loading, the voice/fax channel indicators will cycle from left to right. When all bytes of the file have been transferred, you will receive the following message:

```
DOWNLOAD COMPLETE
DISCONNECTED
```

Glossary

attenuation

Decrease or deterioration of signal strength. Within the voice/fax channel, attenuation is a configuration option for output signal levels in 1–dB increments.

bar graph

A visual indication of the input signal level strength in dB, as displayed on the command port or by the three red indicators on the voice/fax channel.

bit

Contraction of binary digit. The smallest unit of information and the basic unit in digital data communications. A bit can have a zero or a one value (mark or space in data communications terminology).

buffer

A temporary storage area to absorb the difference in either the rate of data flow or the time of occurrence of events in transmissions from one device to another.

busyout

A configuration option that places the voice/fax channel into the busy state, effectively disabling the channel.

call inhibit

A configuration option that prevents the voice/fax channel from calling another voice/fax channel.

ClearVoice Technology

A group of BLACK BOX voice technologies centered around the ITU G.729 voice algorithm. Also includes voice switching, fax/modem demodulation, silence suppression, and background noise regeneration. See G.729.

dB, decibel

A relative measure (logarithmic) of change in signal power.

dBm

An absolute measure of signal power, where 0 dBm equals 1 milliwatt into 600 ohms.

dc signalling

A configurable option in an E&M-strapped voice/fax channel. When so configured, the channel senses the idle/active status by the dc condition of the M-lead, and indicates the idle/active status to the remote end by the dc condition on the E-lead.

decading pulsing

See pulse dialing

DID

Direct Inward Dialing, a service offered by the telephone company. It allows an outside caller to dial an internal extension without passing through the operator. Billing does not start until the extension answers.

DISA

Direct Inward System Access, a service offered by a PBX. It allows incoming calls to the PBX to have dialing access within the private network. May often be protected by a dialed password. Billing starts at the time when the PBX provides the dial tone.

disconnect supervision

A voice communication protocol. It indicates to the local user that the remote end has gone off-hook. This protocol is significant in loop start, where disconnect is denoted by removal of power to the station equipment.

DIT

Direct In Termination, a service offered by a PBX. It allows incoming calls to the PBX to be routed directly to a selected telephone or group of telephones without operator intervention. Billing does not start until the telephone answers.

DTMF

Dual Tone Multifrequency, used for call addressing in pushbutton telephones. Also known as Multifrequency Push-button (MFPB) in Europe.

E&M

In industry usage, a signalling convention between voice PBXs. As related to the voice/fax channel, a strapping option for compatibility with the tie-trunk side of a voice PBX. There are three types (Type I, Type II, and Type V) of E&M strapping options supported by the 3K voice/fac channel.

fast busy signal

See Reorder

fax, facsimile

Transmission of hardcopy images (written, typed, or drawn material) through the voice/fax channel.

fax sharing device

A device used when a fax machine is shared between the Public Switched Telephone Network (PSTN) and the Integration Network. On incoming calls, the fax signals are routed automatically from either the PSTN or the Integration Network to the fax machine. On outgoing calls, the fax machine is selected by the user to the desired destination network.

force connect(ed)

A dedicated connection between any two voice/fax channels. Contrast with switching.

G.729

A voice compression algorithm developed for transporting high-quality voice at an 8 Kbps rate. Also defined as a Conjugate-Structured Algebraic-Code-Excited Linear-Predictive Algorithm.

host

The central computer (or computer systems) that provides primary data processing functions or data base access.

Hz, Hertz

A measure of frequency or bandwidth equal to one cycle per second.

input level

A level of relative analog signal strength obtained from the attached telephone equipment, as measured in the voice/fax channel.

input level display

A visual indication of the input level as displayed by the voice/fax module. Used as a diagnostic tool to monitor the input signal level.

interrupted ring

A configurable option for FXS-strapped voice/fax channels. Causes the local telephone to ring when the remote telephone is lifted off-hook, or vice versa.

KTS

See FXS

line sharing device

A device with the ability to differentiate between voice and fax signals, and used when voice and fax share the same channel. It routes the output of the voice/fax channel either to the telephone interface or to the fax machine, as appropriate.

link

A communications circuit or transmission path connecting multiple points in a network, a composite. There are two types of link in a DVI server:

- An Interconnect Link, that connects two Integration Units, using an analog or digital line provided by the telephone company. An analog line requires a modem at each end of the link. A digital line requires a DSU/CSU at each end.
- A Mux Link, (leased line) that connects an Integration Unit to a feeder BLACK BOX multiplexor.

loopback

A diagnostic feature used to test the voice/fax channel from end-to-end over the link and back.

number of rings

A configuration option for FXS-strapped switching voice/fax channels. On outgoing calls, specifies the number of ringbacks before receipt of a second dial tone.

off-hook

A line condition caused when a telephone handset is removed from its cradle, if active.

on-hook

A line condition that exists when a telephone handset is resting in its cradle, if active.

OPX

See FXO.

PBX, private branch exchange

A privately owned phone system installed within the premises of an organization. It allows communication among users within the organization, as well as between those users and the outside world. It differs from a key telephone system, in that the user must dial an access number such as 9 to get an outside line.

pulse dialing

Method used for call addressing in rotary telephones by short pulses of on-hook/off-hook. Also known as decadic pulsing in Europe.

pulsed E&M

A variation on the E&M signalling format, used in France. A call connect request is made by sending a pulse of a medium duration on the transmit or "Tron" lead, and the call request is acknowledged by a return pulse of the same duration on the receive or "Ron" lead.

receive inhibit

A configuration option that prevents the voice/fax channel from receiving calls.

repeated ring

A configuration option for an FXS-strapped voice/fax channel. When so configured, causes the local voice/fax channel to ring the attached telephone set in response to a remote PBX ringing signal. self-test

A diagnostic feature used to test the voice/fax channel locally that does not include the link.

signalling

A handshaking protocol used between telephone equipment. Includes supervising (on/off hook line status), alerting (ringing), and call addressing (dialing) for switched services.

signalling options

Options related to signalling that are configurable in the voice/fax channel.

When strapped for FXS, the options include repeated ringing, and interrupted ring. When strapped for E&M, the options include DC, 2280 TONE, and PULSED DC.

single frequency

See tone signalling.

slow busy signal

- In a switching mode, the response to a call attempt when the called extension is busy.
- In a force-connect mode, the signal heard when a synch loss occurs (link goes down).

strapping

A hard-wire method used to configure a voice/fax channel. Strappable options include FXS, FXO, and E&M telephone interfaces, and Types I, II, and V of the E&M-strapped telephone interfaces.

switching

Method of connecting any two voice/fax channels by dialing the extension number of the destination channel. Not applicable to force-connected voice/fax channels; voice/fax channels that are force-connected cannot dial out nor can they receive calls.

telephone interface

The analog side of a voice/fax channel. Compatible with FXS, FXO, or E&M type telephone interface equipment, as strapped.

telephone interface connector

A termination on the voice/fax module that connects the analog side of the voice/fax channel to the telephone equipment such as a telephone set, key telephone system, or voice PBX. There are two types of telephone interface connectors: RJ11 and RJ45.

tone signalling

A configurable option in E&M-strapped voice/fax channels. When so configured, the local channel senses the idle condition of the remote channel when a 2280-Hz tone is present on the T1-R1 pair. When the local channel is in an idle state, it informs the remote channel by placing a 2280-Hz tone on the T-R pair.

Types I through V

Strapping options, used to match the E&M-strapped voice/fax channel with one of five PBX trunk types. See Appendix B for schematic details.

2-wire/4-wire

A configuration option that matches the E&M-strapped voice/fax channel with the PBX tie trunk. In a 4-wire system, the audio signal is transmitted over one pair of wires and received over the other pair. In a 2-wire system, the same pair of wires is used for both transmit and receive.

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